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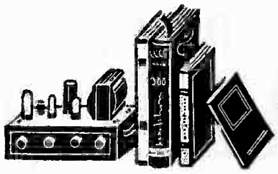

audiophile's bookshelf

Three recent books by three experts

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By JOSEPH MARSHALL. A much needed book on the specialized approach necessary to service high-fidelity equipment. The author discusses not only electronic faults, but the mechanical and acoustical defects which cause a hi-fi system to function less than perfectly. The book is clearly written and well indexed. Hard cover, $5.00 **232** Soft, $2.90 **233**

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By CHARLES FOWLER, Publisher, AUDIOCRAFT Magazine

Whether your budget is small or large, whether you are a beginner or a devotee from way back, this complete and definitive guide can help you get the most enjoyment out of your high-fidelity system.

Here is the practical expert advice needed by everyone who has, or expects to have, a high-fidelity system. Starting with an introduction to sound, the author then describes the various links in the hi-fi chain, explaining their features and specifications so as to provide the most helpful criteria for evaluating, and for selecting and matching components. $4.95 **234**

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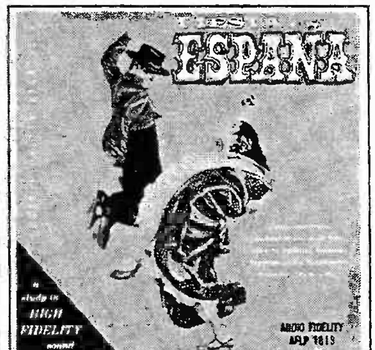

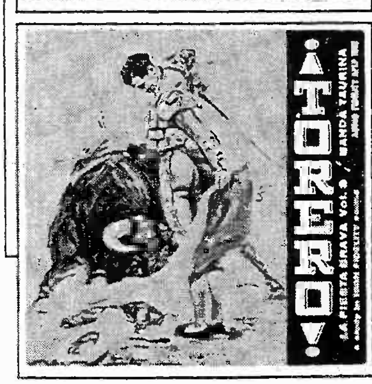

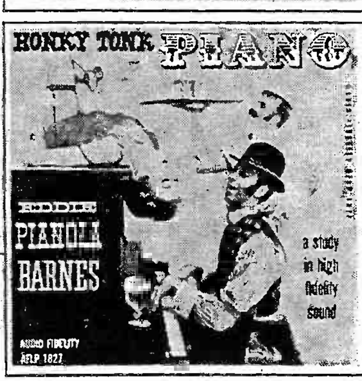

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The Grounded Ear

by Joseph Marshall

Reducing Distortion

High fidelity is achieved not so much by revolutionary circuitry as by patient efforts to refine conventional circuits. Sometimes a very small thing can make an unexpectedly large difference in performance. A notable recent example of this is a circuit refinement discovered by David Hafler, which is being incorporated into the latest shipments of the Dynakit amplifier to improve its already fine performance.

The Dynakit amplifier uses EL34's with fixed bias in the output stage. Hafler reports that a small resistor (12 ohms appears to be the optimum size for this application), inserted in the common cathode circuit (Fig. 1), reduces distortion appreciably. The small common resistor appears to have the effect of correcting for imbalance between the output tubes. If the output tubes are not perfectly balanced, second-harmonic distortion is not completely canceled out. When this residue of un-

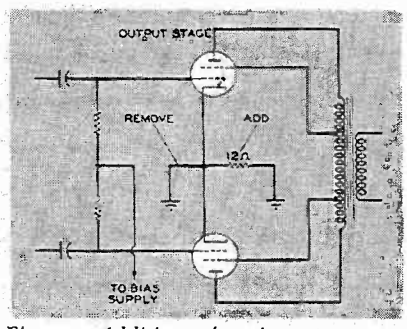


Fig. 1. Addition of resistor to output stage of Dynakit improves performance.

balanced distortion appears in the common cathode resistor, the phase relationship of the second-harmonic distortion is correct to produce negative feedback resulting in the reduction of distortion. The imbalance can be corrected in various other ways, but all require some means of indicating the point of balance. The Hafler trick has the virtue of doing the job automatically and without the need for meters, scopes, etc.

Although developed for the Dynakit, the device should also work with other amplifiers. The value of the resistor is related to the transconductance of the tubes; the higher the transconductance, the higher the value of the resistor. The range is from 5 to about 25 ohms. Where the tubes operate with fixed bias, the resistor is inserted between the two cathodes and ground. It will also work in self-bias output stages, where it is inserted on either end of the present, bypassed cathode resistor. The gimmick works best with ultra-linear or triode operation where the odd-order distortion is very small, and the largest component of distortion is second harmonic due to incomplete balance. However, the device is so simple and inexpensive that it is worth trying with almost any design or output tubes.

New Cook Test Record

It has been quite a while since we have had a test record offering any real challenge to the capabilities of cartridges, arms, amplifiers, and speakers. In the last few years improvements in these components have so far outstripped test material that there has been a tendency toward the comforting belief that we were fast approaching ideal performance. Emory Cook's new Series 60 *Chromatic Scale Test Record* is going to come as a shock to quite a few people. Portions of it provide so severe a challenge that owners of even the finest available components, especially cartridges, arms, and speakers, are likely to be unhappy about the performance of their pride and joy.

The Series 60 has several unique features. For one, it offers test tones in the chromatic musical scale rather than the arbitrary mathematical scale used hitherto. For another, one side of it is recorded with Fletcher-Munson compensation. These innovations will be useful in many ways, but it is in the bands providing tone bursts for testing transient response and the series of rising-amplitude tones for checking dynamic range that the recording poses a real challenge to present standards of performance.

Particularly demanding are the series of low-frequency tone bursts on the Fletcher-Munson side. There are three sets of these at 61, 92, and 123 cps. In each set there are three trains of bursts — the first is two vibrations long; the second, four vibrations; and the third, eight vibrations. The bursts are not only very rapid, but are also recorded with a very big swing. I rather doubt that even Emory Cook owns a system that can pass this test with flying colors. The "Comparative Intensity Level" band on the Fletcher-Munson side presents, in the last of the three rising levels of the 32-cps tone, about as big a swing as can be found on records, and the arm and pickup that will reproduce it — or even track it — deserve some kind of medal. The same series are repeated on the B side which is recorded with conventional NAB bass de-emphasis. The much lower swing presents a far less challenging test, but equipment that can pass it will deserve warm praise.

Some people may excuse the failure of components to meet the demands of the Series 60 by contending that the tone bursts, especially, are more severe than any presented by musical material, and that such components may still be capable of doing justice to musical material. There may be truth in this contention. On the other hand, today we are taking for granted performance considered impossible two or three years ago.

A long time ago Emory Cook presented a similar challenge to disc reproduction with his original, and then incredible, 20-to-20,000-cps test record. It would be absurd to say that this record was responsible for the revolution in disc reproduction we have enjoyed since then; but unquestionably it helped. With the Series 60, Emory Cook again throws the gauntlet in the face of accepted standards of performance. I feel sure that some workers in the hi-fi field will rise to the challenge and that two years from now we will have components which will demand even more severe material to try their mettle. Meanwhile, Cook is certainly to be congratulated, not only for so successfully raising the bar of performance, but for developing techniques which made it possible to record it. Obviously, the record presents a standard, both for reproducing equipment and, perhaps even more, for recording equipment.

The Freel Speaker

Some months ago I commented on a paper in the British *Wireless World* reporting a new configuration for electrostatic speakers by Leak and Sarkar which, the authors claimed, would result in better and cheaper electrostatic speakers. Mr. Leak (of Leak amplifier fame) has for the past two months carried a box in his advertisements announcing that the *Freel* speaker embodying this new design would be introduced some time this year.

Demonstration Tapes

A few months ago I also commented on the fact that there was a need for demonstration tapes capable of showing off the superiority of tape over first-class-disc reproduction. E. D. Nunn, whose Audio-

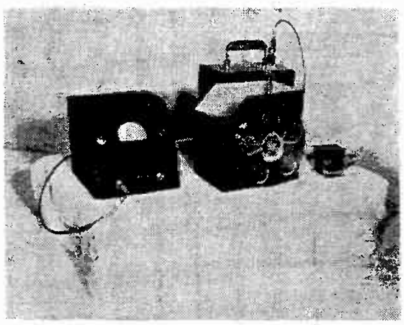
Continued on page 48

Audionews

PROTECTIVE CANS FOR RECORDING TAPE

Laboratory tests show that unprotected tape recordings may be damaged or destroyed by commonplace electromagnetic fields. Electric motors on appliances such as vacuum cleaners and air conditioners, as well as generators, transformers, etc., may be the cause of damage.

Cans of *Fernetic-Co-Netic*, lightweight, double metal construction will protect tape recordings against electro-



Can shields tape from magnetic fields.

magnetic fields of all intensities. These cans, manufactured by the Magnetic Shield Division of the Perfection Mica Company, are available in several sizes. The largest size accommodates 24 10½-inch reels.

Additional information about Fernetic-Co-Netic tape cans will be furnished on request.

FAIRCHILD BOOKLET

Fairchild Recording Equipment Company has published a booklet entitled *How Good Is Your Arm.* The booklet covers such aspects of arm design as resonance, tracking, tracking error, torsional resonance, and pivot design. It is available without charge on request.

PLASTIC LEADER TAPE

A new plastic leader and timing tape featuring an antistatic coating and a reported 50% increase in strength has been introduced by Minnesota Mining and Manufacturing Co. for use with magnetic recording tape.

The antistatic coating is said to reduce noise as the tape passes over the playback head, as well as making the tape easier to handle.

To be spliced to the beginning and end of a reel of magnetic tape, the new nonmagnetic tape provides a protective leader that makes for easier tape threading and prevents damage to the recorded material.

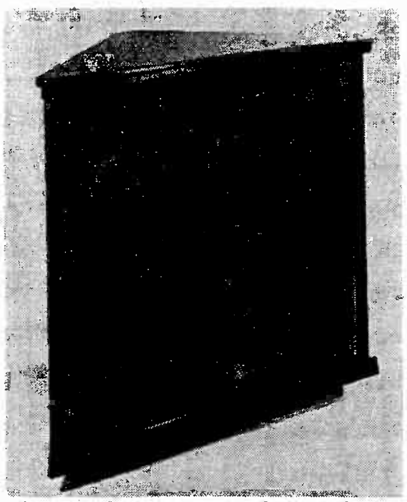
As a timing tape, it can be inserted between recorded selections to provide "dead air" or to facilitate cuing. It can be labeled or marked with either a pencil or a ball-point pen.

The tape is printed with indicator arrows every 15 in., and Scotch plaid sections are printed every 7½ in. Scotch brand plastic leader and timing tape is ¼ in. wide by 150 ft. long. It comes in a special self-dispensing box, and is available through all dealers who sell Scotch brand magnetic tape.

NEW KNIGHT SPEAKER ENCLOSURES

Allied Radio Corporation has announced the release of two new Knight speaker enclosures in kit and completed form. Two models are available; corner enclosure and bass-reflex enclosure.

Each enclosure can be used with a 12-inch or 15-inch speaker for single, 2-way, or 3-way speaker systems. A precut adapter board for the addition of



The Knight corner speaker enclosure.

tweeters or other speaker components is supplied with each enclosure. Both cabinets are constructed of ¾-inch plywood with mahogany veneer. Plastic grille cloth is included.

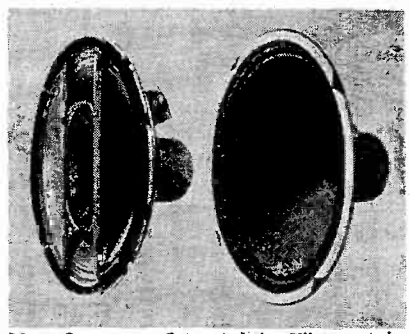
The corner enclosure is available as a prefinished (in French mahogany) build-it-yourself kit; unfinished build-it-yourself kit; and completely assembled, factory-finished in French mahogany.

The bass-reflex enclosure is available as an unfinished build-it-yourself kit; unfinished, completely assembled; and assembled, factory-finished in French mahogany. All models of this unit are supplied with legs; the enclosure can be stood upright in highboy style, or laid on its side in lowboy style.

Builder's kits include parts, screws, glue, and step-by-step instructions for quick, easy assembly. Unfinished kits are smoothly sanded, ready for finishing. Prices range from $24 to $69.

SONOTONE SPEAKERS

The Sonotone Corporation has introduced the *CA-12*, a 12-inch coaxial loudspeaker. The CA-12 provides high flux density (woofer, 12,000 gauss; tweeter, 8,500 gauss), and is said to cover the frequency range from 40 to



New Sonotone CA-12, left; W-12, right.

14,000 cps. An elliptical cone tweeter is used for improved dispersion of high frequencies. Net price of the CA-12 is $19.50.

The Sonotone W-12 12-inch woofer is the same low-frequency unit used in the CA-12. The W-12 is priced at $12.00.

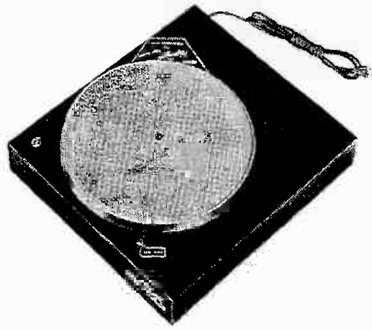
MAGNECORD CATALOGUE

Magnecord, Inc., recently announced the publication of a new catalogue of its entire line of professional magnetic tape recorders. Included are complete specifications and photographs of all models, as well as full information about accessories and modification kits.

The catalogue is available free on request.

DUO-SPEED TURNTABLE

The *Duo-Speed Professional* turntable, manufacturerd by Components Corporation, features a positive-action speed-control lever and a belt-driven, heavy-weight, nonmagnetic 12-inch turntable.



The Duo-Speed turntable is belt-driven.

The unit is powered by a 4-pole, constant-speed motor that is magnetically shielded.

The Duo-Speed Professional is available in two models; the Model 45 for 33⅓ and 45 rpm, and Model 78 for 33⅓ and 78 rpm. Both models retail for $49.50. A hand-rubbed walnut base for the turntable is available at extra cost.

REK-O-KUT BOOKLET

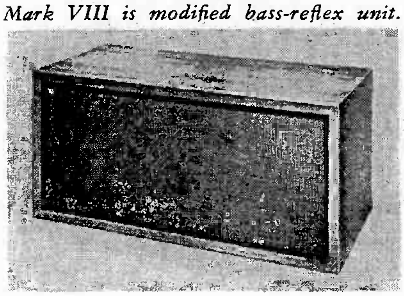
Turntable or Record Changer . . . Which Shall I Buy? is the title of a booklet issued by the Rek-O-Kut Company, Inc. The purpose of the booklet is to simplify the essential differences between the turntable and the record changer. *Turntable or Record Changer* is available free on request.

AMERICAN LOUDSPEAKER HI-FI LINE

The American Loudspeaker Division of Contemporary American Furniture, Inc., has just introduced a new line of hi-fi products. Initially, the new line includes two matched speaker systems and an equipment console. All three units are available in three finishes: traditional mahogany, sandalwood mahogany, and natural walnut.

The *Mark VIII* speaker system is a modified bass-reflex unit containing an 8-inch bass driver and a 3½-inch cone tweeter. Response of the system is said to be essentially flat from 60 to 12,000

Mark VIII is modified bass-reflex unit.

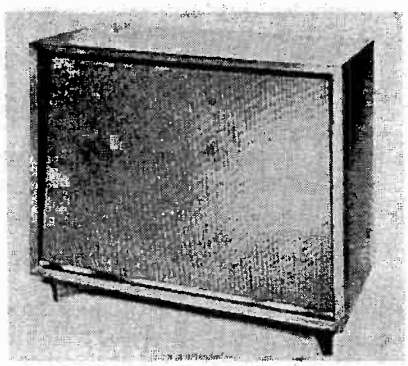


cps. Cabinet walls and back are ¾ in. thick, and each mitered joint is secured with glue and a metal spline. A grid formed by 12 routed slots in the front mounting panel provides the desired loading of the port.

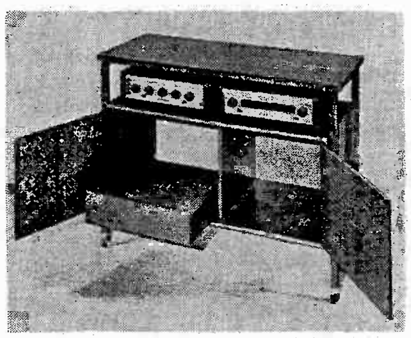
The system's dimensions are 24 in. wide by 10 in. high and 12 in. deep. The Mark VIII is available in mahogany at $49.50, and in walnut and sandalwood at $52.50.

The *Mark XII* is a two-way speaker system employing a 12-inch bass driver and a 5-inch cone tweeter. Cabinet walls are ¾ in. thick.

Port loading is accomplished by a slotted grid coupled to a flared duct formed by the cabinet base. Speakers are secured to the face panel and sealed.



American's 2-way Mark XII speaker system is shown above. Below is Mark 100 equipment cabinet for hi-fi components.



Dimensions of the system are 27 in. wide by 23 in. high and 14 in. deep. Response is advertised as being essentially flat from 50 cps to audible limits.

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

The Mark XII is available in mahogany for $97.50, or in walnut and sandalwood for $105.00

The *Mark 100* equipment console was designed to provide a cabinet for high-fidelity components. The unit houses a complete audio system, except for speakers, and stores 100 records. The Mark 100 includes solid brass ferrules for the legs, brass piano hinges for the doors, magnetic door catches, ball-bearing drawer slides for the changer, and a precut changer drawer. The lacquered hardboard back is predrilled to make installation of equipment easy.

Dimensions of the Mark 100 are 36 in. wide by 27 in. high and 16 in. deep. The unit is available in mahogany finish for $97.50, and in walnut and sandalwood for $105.00.

QUICK-THREAD TAPE REEL

A new *Quick-thread* reel designed to provide easier, faster loading of recording tape is now regularly supplied with 5- and 7-inch reels of Reeves Soundcraft tapes. The new reel features a loading slot accessible at the outer edge of the reel. Tape is inserted into this slot and automatically guided to the hub for quick, secure attachment.

Each side of the Soundcraft "Quick-thread" reel has an indexing area with a special surface to permit indexing of the reel with all types of pens and pencils.

TWO NEW ACCESSORIES

The Dynamic Hi-3 speaker selector *Model DS77*, manufactured by Dynamic Electronics — New York, Inc., is designed for temporary or permanent speaker coupling and comparison checks. It may be used in connection with any audio output for switching to not more

One of two new accessories by Dynamic.

than three speakers at remote points. The selector comes in a compact, all-metal case with three-position switch.

Put out by the same manufacturer is the Dynamic three-way antenna selector *Model DA88*. This accessory is intended for use in cases where up to three individual antennas are to be coupled to one receiver, as in mixed UHF and VHF areas or where high- and low-channel reception requires antenna variation. Three lead-in wires can be connected to the selector (either 300 or 75 ohm); then one set of leads is connected from the selector to the TV, FM, short-wave, or any multi-band receiver.

TIPS FOR THE WOODCRAFTER

by George Bowe

Building In, Part II

Last month we examined the skeleton of a frame house, and this month we prepare for surgery on the house itself. Before we lift the scalpel — or the wrecking bar — we must have a pretty good idea of what we expect our built-in to be. We have a choice of building in equipment alone or of constructing a combination of equipment and bookcase, storage wall, new partition, or counter. Two combination installations are shown in Fig. 1. While a built-in is generally something actually built into part of the house structure, the term is frequently used in reference to items merely set against a wall or fastened to it. If your project is to be of the latter type, you may stop reading right now and begin construction. If, however, your plan calls for digging into an existing partition, stay with us, for we have more to say about that.

Speaking of a plan, it's an excellent idea to make a working drawing as complete as possible before starting — at least a front elevation with dimensions. If you are adapting your project from a photograph of a well-designed unit, retain the proportions as closely as possible as you adapt it to fit the area in your home. If you don't, your reproduction may not bear a very strong resemblance to the attractive original.

Naturally, location of the installation is extremely important and should relate to its use: speakers should be positioned for most effective listening, movie screens and TV for best viewing. Equipment such as tuners, turntables, and projectors should be planned for convenience and ease of operation. In deciding on the location, consider also furniture arrangement in the room. An improperly placed built-in might necessitate an awkward furniture grouping in order to utilize properly the facilities of the installation. If a heavy piece of furniture must be moved each time the built-in is used, much of the enjoyment of the equipment is lost, and its value diminishes.

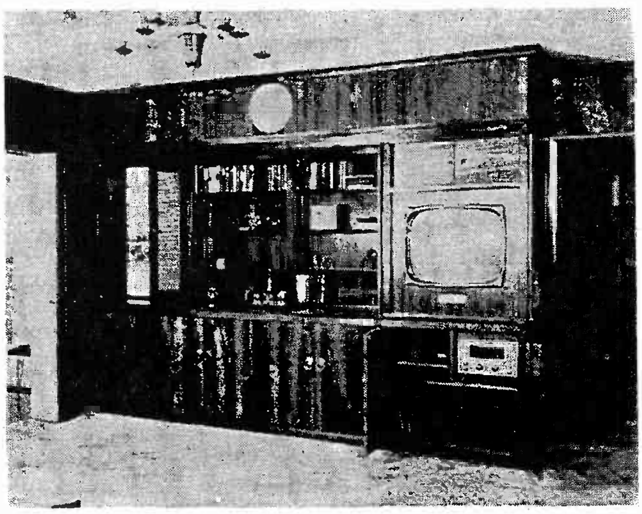
When a location for the installation has been selected, check it to determine whether the built-in will affect the operation of doors, windows, wall switches and outlets, light fixtures, and radiators or heat registers. Large wall areas usually present few problems but, where space is precious — in hallways, small rooms, etc. — it pays to plan very carefully to avoid annoying errors. Most times electric switches and outlets can be moved without difficulty to accommodate the new construction. Where heating or plumbing systems are affected, it is wise to call a professional in advance to determine how to reroute that section of the duct work or piping.

Since the average home wall is constructed of 2-by-4 studding, the depth of the wall itself is not sufficient to house completely even a bookcase. To overcome this limitation, the front of the built-in can extend into the room, or the back of it can project into a room at the rear or into a closet or an area under stairs. In the event that the back is exposed in another room, it can be stained and treated as a panel or it can be painted or papered. It might also lend itself to holding a bulletin board or to mounting photographs or paintings. But regardless of whether the installation projects to the front or rear, make its depth ample enough to contain comfortably the equipment and all other items it will be expected to hold. If it is to be a floor-to-ceiling arrangement for equipment and books, the upper section can be shelves of 1-by-8 or 1-by-10 lumber to take the books and smaller equipment, while the lower portion can be cabinets of greater depth to house larger components such as loudspeakers and a turntable.

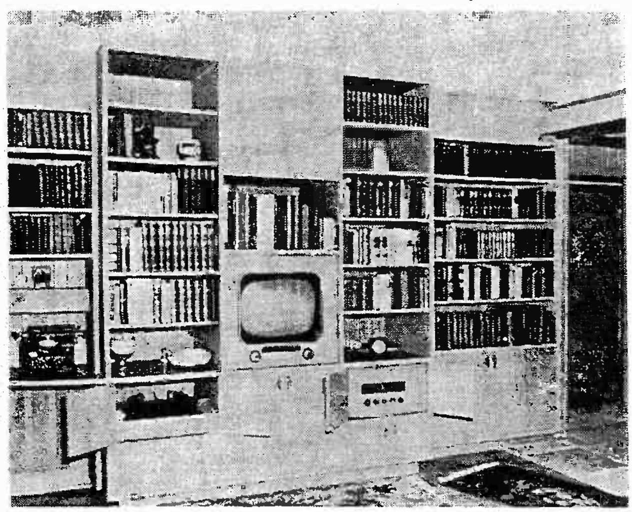
As we mentioned in the May article, it must be determined in advance whether the partition to be operated upon is bearing or nonbearing. A bearing partition supports an overhead load; a nonbearing partition serves only as a room enclosure and supports nothing above it. Last month we explained how to tell if a partition is bearing or nonbearing. Usually a bearing partition runs at right angles to the ceiling joists and supports them, while a nonbearing partition runs in the same direction as the ceiling joists.

Continued on page 44

Figs. 1A and 1B. Two examples of successful built-in sound installations. Each includes TV, radio, and phono facilities.



Courtesy Conrac, Inc.



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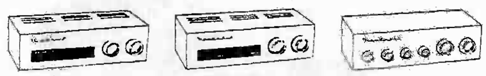

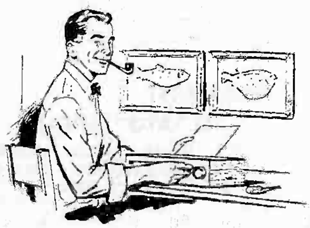

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❸ **HEATHKIT HIGH FIDELITY PREAMPLIFIER KIT** This preamplifier meets or exceeds specifications for even the most rigorous high fidelity applications. It provides a total of 5 inputs, each with individual level controls. Hum and noise are extremely low, with special balance control for absolute minimum hum level. Tone controls provide 18 db boost and 12 db cut at 50 cps, and 15 db boost and 20 db cut at 15,000 cps. Four-position turnover and four-position rolloff controls for "LP", "RIAA", "AES", and "early 78" equalization. Derives power from main amplifier, requiring only 6.3 VAC at 1A and 300 VDC at 10MA. Beautiful satin-gold enamel finish. Shpg. Wt. 7 lbs.

MODEL WA-P2 (with cab.) **$19.75**
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❹ **HEATHKIT ADVANCED-DESIGN HI-FI AMPLIFIER KIT** This fine 25-watt high fidelity amplifier employs KT66 output tubes by Genalex and a Peerless output transformer for top performance. Frequency response ± 1 db from 5 to 160,000 cps at 1 watt. Harmonic distortion less than 1% at 25 watts, an IM distortion less than 1% at 20 watts. Hum and noise are 99 db below 25 watts. Output impedance is 4, 8 or 16 ohms. Extremely stable circuit with "extra" features.

MODEL W-5: Consists of W-5M *plus* WA-P2 Preamplifier
Shpg. Wt. 38 lbs. Express only **$79.50** $7.95 dwn. $6.68 mo.

MODEL W-5M **$59.75** $5.98 dwn. $5.02 mo.
Shpg. Wt. 31 lbs. Express only

❺ **HEATHKIT DUAL-CHASSIS HI-FI AMPLIFIER KIT** This 20-watt Williamson-type amplifier employs the famous Acrosound model TO-300 output transformer, and uses 5881 tubes. Frequency response is ± 1 db from 6 cps to 150 kc at 1 watt. Harmonic distortion less than 1% at 21 watts, and IM distortion less than 1.3% at 20 watts. Output impedance is 4, 8 or 16 ohms. Hum and noise are 88 db below 20 watts.

MODEL W-3: Consists of W-3M *plus* WA-P2 Preamplifier
Shpg. Wt. 37 lbs. Express only **$69.50** $6.95 dwn. $5.84 mo.

MODEL W-3M **$49.75** $4.98 dwn. $4.18 mo.
Shpg. Wt. 29 lbs. Express only

❻ **HEATHKIT SINGLE-CHASSIS HI-FI AMPLIFIER KIT** This 20-watt Williamson-type amplifier combines high performance with economy. Employs Chicago-Standard output transformer and 5881 tubes. Frequency response ± 1 db from 10 cps to 100 kc at 1 watt. Harmonic distortion less than 1.5% and IM distortion less than 2.7% at full output. Output 4, 8 or 16 ohms. Hum and noise—95 db below 20 watts.

MODEL W-4A: Consists of W-4AM *plus* WA-P2 Preamplifier
Shpg. Wt. 35 lbs. Express only **$59.50** $5.95 dwn. $5.00 mo.

MODEL W-4AM **$39.75** $3.98 dwn. $3.34 mo.
Shpg. Wt. 28 lbs. Express only

❼ **HEATHKIT 20-WATT HIGH FIDELITY AMPLIFIER KIT** Features full 20 watt output using push-pull 6L6 tubes. Built-in preamplifier provides four separate inputs. Separate bass and treble controls. Output transformer tapped at 4, 8, 16 and 500 ohms. Designed for home use, but also fine for public address work. Response is ± 1 db from 20 to 20,000 cps. Harmonic distortion less than 1% at 3 db below rated output. Shpg. Wt. 23 lbs.

MODEL A-9B **$35.50**
$3.55 dwn., $2.98 mo.

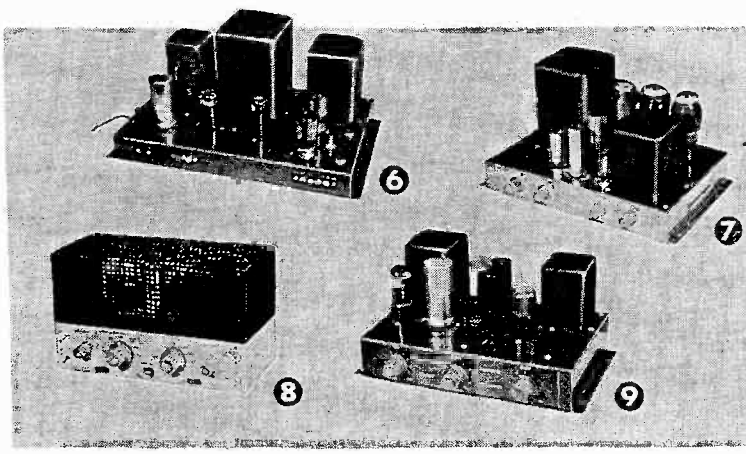
❽ **HEATHKIT ELECTRONIC CROSS-OVER KIT** This device separates high and low frequencies electronically, so they may be fed through two separate amplifiers driving separate speakers. Eliminates the need for conventional cross-over. Selectable cross-over frequencies are 100, 200, 400, 700, 1200, 2000 and 3500 cps. Separate level controls for high and low frequency channels. Attenuation 12 db per octave. Shpg. Wt. 6 lbs.

MODEL XO-1 **$18.95** $1.90 dwn., $1.59 mo.

❾ **HEATHKIT 7-WATT ECONOMY AMPLIFIER KIT** Qualifies for high fidelity even though more limited in power than other Heathkit models. Frequency response is ± 1½ db from 20 to 20,000 cps. Push-pull output and separate bass and treble tone controls. Good high fidelity at minimum cost. Uses special tapped-screen output transformer.

MODEL A-7E: Same as A-7D except one more tube added for extra preamplification. Two inputs, RIAA compensation and extra gain.
Shpg. Wt. 10 lbs. Incl. Excise Tax **$19.95** $2.00 dwn. $1.68 mo.

MODEL A-7D **$17.95** $1.80 dwn. $1.51 mo.
Incl. Excise Tax
Shpg. Wt. 10 lbs.



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TAPE | NEWS & VIEWS

by | J. Gordon Holt

The Basic Recorder

Several months ago I happened to mention a few isolated aspects of tape-recorder theory in connection with the practical matters of equalization and bias adjustment, and these vague references apparently aroused the curiosity of a number of readers. Letters from them have asked, specifically, how tape recording and playback equalization differ from disc equalization; why it isn't necessary to roll off the highs in playback after having boosted them in the recording process; how the requisite sharp recording equalization curve is obtained; and why it is necessary to use ultrasonic bias in tape recording.

So, a few words of explanation are in order. Anyone who already knows the answers to the questions above may be excused from class this month.

First, let's take a look at the medium itself — the magnetic tape coating. This is a layer of fine-grained iron oxide which exhibits much the same magnetic properties as any other ferrous material. It is essentially a nonlinear recording medium, which is to say that if we try recording an audio signal on it directly we get a grossly distorted recording. The effect is caused by hysteresis: residual magnetization of the oxide after the magnetizing force is removed. Distortion from this effect is limited to an operating region close to zero magnetization on each side (plus and minus) of zero. If an ultrasonic tone of constant frequency (the *bias*) is superimposed on the audio tone to be recorded, the *average* magnetization of the oxide becomes very nearly proportional to the audio wave form — which is the effect desired. There isn't complete agreement as to exactly how this can best be explained physically; but this ultrasonic recording bias, mixed with the audio signal, does reduce tape distortion to a reasonable level.

The bias frequency is set at 30,000 cps or higher, and goes as high as 200,000 cps in some professional tape recorders. It should be at least 4 times that of the highest tone to be recorded, and the higher the better. Higher frequencies reduce the possibility of beat notes between the bias tone and the harmonics of the audio signal, and seem to improve the definition and transparency of the recorded sound.

The wave length of the bias frequency is small compared to the magnetic gap width in the erase head; that is, several complete bias-frequency cycles occur while any point on the recording tape is moving across the gap. Therefore the average magnetization of the bias frequency alone is zero, and it leaves the tape demagnetized or erased. It is used for that purpose as well as in recording. While a very high bias frequency is desirable in recording, it is a mixed blessing, because the higher the frequency, the more current is required through an erase head for complete erasure. Too much current through the erase head heats it up, making it adhere

to the tape. So we compromise, simply because it makes sense to use the same oscillator for erasure and biasing.

So much for the economic aspects of tape biasing. The second characteristic of tape as a recording medium, its equalization requirements, is rather difficult to visualize, because it is dependent simultaneously on the tape, the record head, and the playback head.

Disc equalization is no problem to understand, because if a certain recording curve is used, consisting of a certain amount of treble boost and a certain amount of bass attenuation, this same curve will reappear when the disc is played with a good magnetic pickup. So if we want flat response from a disc, we simply feed a flat signal to the record amplifier. Not so with tape, however.

If an unequalized signal were fed to a tape head and then played back without equalization, the resulting curve would resemble a mountain peak in cross section. There would be a broad hump somewhere above the middle of the audio range, and rapidly collapsing response above and below that region: the ultimate in presence peaks! The top of this peak occurs at a frequency determined by tape speed (the higher the speed, the higher the frequency) and head gap width (the narrower the gap, the higher the frequency). For ¼-mil heads at 7½ ips, the maximum unequalized response occurs at about 3,000 cps.

To get a *flat* response from tape, then, we have to boost bass and treble during the recording or playback process. Remember, this is necessary just to get a flat response from the tape, and does not have anything to do with the equalization used on discs to improve their signal-to-noise ratio. At no point in the process is bass cut or treble rolloff required; the tape head's inherent response characteristics take care of these.

Published charts showing the energy distribution at certain frequencies in music and natural sounds indicate that there is almost as much power present in the bass range as in the middle range, but above 3,000 cps or so (the overtone range) the energy distribution falls off rapidly. A second factor to consider in tape equalization is that, while good design can reduce hum to a negligible level, nothing can reduce tape hiss to much below a certain prescribed level. Juggling these observations around a bit, it becomes apparent that if we try to boost bass while recording, we will have to depress the over-all recording level to prevent tape saturation at low frequencies, and will lose signal-to-noise ratio. However, since hum does not necessarily have to be a serious problem in playback, we can easily do all the bass boosting then.

What about the highs? Well, the energy-distribution curves show that we can boost them pretty heavily while recording without having to reduce our over-all recording level, but if we were

Continued on page 46

by RICHARD D. KELLER

book

How to Use a Tape Recorder

Dick Hodgson and H. Jay Bullen; pub. by Hastings House, New York; 216 pages; $4.95.

Pamphlets and booklets have appeared in the past listing multitudinous uses for tape recorders, but none can match this volume in completeness. Not only does it list practical uses for the tape recorder in business and the home, but it delves into the preparation and forethought that is necessary to make tape recording the effective tool it can be. After discussing the many advantages of tape over older methods of sound recording, and giving the history of Danish, German, and American developments in the tape field, the authors point up some pertinent suggestions for anyone making recordings: such things as how to organize thoughts effectively before recording, and the development of selective listening during playback.

The largest part of the book gives suggestions for using a recorder in just about every phase of business from selling to safety, and it is here that almost every reader will find some application of tape recording to his own business world. Among nonbusiness uses in the next section he will find suggestions for parties that will have him warming up the recorder before his next guests arrive.

It should be emphasized that the book does *not* discuss microphone techniques or problems of acoustics; it was written to show the recorder owner how to get the greatest possible use from a versatile modern tool of communication and sound preservation.

When it comes to advice on selecting a recorder, the authors' novel approach is to give complete specifications for an ideal (nonexistent) portable recorder, and then to go down the list, discussing each specification and the alternatives that are available. Final sections on accessories and a glossary of technical terms round out one of the best books yet on tape recorders and their applications.

Inverse Feedback

Ed. by Alexander Schure; pub. by John F. Rider, Inc., New York; 48 pages; $0.90, paper bound.

It isn't often that the subject of inverse feedback is presented so that a person

reviews

only casually acquainted with electronic circuits can really understand the basic principles involved. This inexpensive booklet does just that, however. Some simple mathematics is essential to the comprehension of feedback, and just enough is included here to allow the student, technician, or practicing engineer to follow the development of fundamental concepts. More advanced concepts, such as Nyquist diagrams and network theory, are not included.

Clear, descriptive analyses are given of the effects and advantages of negative feedback, types of inverse feedback, its effects on input and output impedances, phase relationships, frequency response, stability, and typical circuit applications. Voltage and current feedback and combinations of them are explained. The book tells how inverse feedback can improve response characteristics and reduce hum, noise, and distortion.

Hi-Fi Equipment Yearbook — 1957

Ed. by Sanford M. Herman; pub. by Herman and Stephens, New York; 128 pages; $1.95, paper bound; $2.75, cloth bound.

An illustrated directory of equipment currently available for high-fidelity music systems, this book contains accurate authorized data, including net prices, on hundreds of items.

It includes all sorts of equipment in all price ranges, from kits to the most expensive factory-assembled components, from phono cartridges to broadcast-quality tape recorders. The information has been furnished by the manufacturers themselves. No attempt has been made to evaluate equipment, but the editors have included accurate specifications for each item. Listings are by category (amplifiers, tuners, etc.), with equipment listed alphabetically by manufacturers within each category. Some manufacturers evidently did not provide information on their products, for a few of the largest and most popular lines of kits and components are not listed.

Perhaps this sort of catalogue is better left to the larger wholesale houses, who handle wide lines of equipment and mail out extremely well-illustrated annual catalogues, plus periodic up-to-date supplements, merely for the asking. Special mention, however, should be made of the cross-reference index, and of the entertaining and informative introduction by Peter Aczel, a New York advertising executive handling hi-fi accounts.

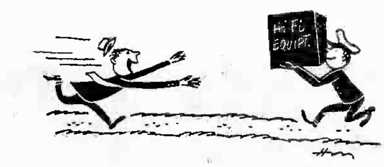
How to Install and Service Intercommunication Systems

Jack Darr; pub. by John F. Rider, Inc., New York; 150 pages; $3.00, paper bound.

Remember the old speaking tubes on ships and in some palatial mansions for communicating between rooms and up and down floors ("decks" to you seagoing enthusiasts)? Well, the science of electronics has brought these systems up to date, and now the sigh of a sleeping baby in the nursery can be heard in the kitchen several rooms away, a business receptionist's sweet tones are conveyed to the boss's remote sanctum, and the Captain's orders are audible over the steam-room roar.

The book attempts to cover all phases of commercial intercommunication work, and to describe design and installation techniques which have been developed during many years of experience by the author. Since transmission of *intelligence* (rather than absolute realism) is involved, justification for the narrow 500-to-3,500-cps voice-channel frequency response of most such systems (including telephones) is established at the outset; and the reason for the general use of combination loudspeaker/microphones of the 45-ohm variety is explained.

The book is divided into chapters covering the essentials of all intercom systems, AC-DC and AC-only systems, cables, installation, servicing techniques, wireless intercoms, designing and building simple intercoms, and unusual applications. Representative commercial types of each classification are presented, complete with schematics and trouble-shooting and trouble-prevention schemes (especially concerning switches and cables).



Readers' Forum

Gentlemen:
How about consructing a 10- to 15-watt amplifier for us — by "us" I mean those of us who are just barely smart enough to count money and measure with a jigger. Use one of David Hafler's circuits: one 12AX7, two EL-84's, and a Stancor PC8405 in the power supply.

Write it up as you did the Huff-Zabriskie 4-watt job, with templates and *pictorial* layout, and don't tell us (see above) that we can get it better, and probably cheaper, ready made.

I haven't had as much fun since the time my father brought home a toy soldering iron, 'way back about 1910. I got real busy and forgot I knew how to solder.

Harry E. Moore
Boonville, Mo.

Such an article isn't of much use unless it does *describe a unit that can't be obtained in a less-expensive kit or commercial unit. These articles are more and more difficult to find. We'll do our best, though, and may be able to come up with such an amplifier soon.* — ED.

Gentlemen:
I seem to have my share of troubles in tuning up my set and keeping it in top operating condition. Is there any book that describes the various tests that can be made on high-fidelity equipment, and tells how a set can be improved? What I am looking for are test methods currently in practice, described under one cover.

Stanley W. Cairns
Pleasantville, N. J.

An excellent book on test procedures for high-fidelity equipment is Maintaining High-Fidelity Equipment, *by Joseph Marshall. The cost is $5.00 for the hard-cover edition, or $2.90 for the soft-cover edition. It is available through most book stores, or it can be ordered directly from the Book Department of* AUDIOCRAFT. — ED.

Gentlemen:
I would like to report that I am well pleased with the results of Mr. Geraci's article on modification of the Pentron PMD-1 tape recorder (AUDIOCRAFT, November 1956, p. 18, "Rebuild your Recorder"). I paid the $45 for the motor and carried out Mr. Geraci's instructions for remodeling the mechanism. I had already made most of the circuit changes, except for installing the Dynamu heads,

Continued on page 48

EDITORIAL

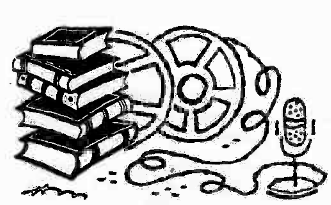
LIKE many another whose profession, as well as hobby, is juggling words, I'm too often guilty of using them hastily, extravagantly, or inaccurately — much like an audiophile who, after laboriously acquiring the finest available home sound system, often employs it for sensationally dramatic, but essentially meaningless, "demonstrations" or carelessly permits the distortion level to rise above tolerable limits. Yet in each case, we are the very ones who should set the standards for the highest fidelity, whether it be to the basic ideas expressed by our words or the music reproduced by our electroacoustical equipment. For not only are the right words and the best sonic qualities vitally important in themselves, but they are the only sure means we have of infecting others with our own enthusiasms and ideals.

I like to think that a large part of the current success of the whole good-sound movement is the consequence not merely of strictly technical developments in circuits and components, but also of the evolution of a specialized *language* for talking, writing, and reading about audio. Most of us at times have cursed vigorously the looseness of a term like "high fidelity," and felt that it is too often abused and misused; yet how helpless we'd all be if we didn't have such a convenient, catchy, and distinctive name for what we're interested in! And while much audio jargon (like any other) may sound exotic — or funny — to the uninitiated, it is just what first arouses their curiosity. Indeed, it is such vivid verbal concoctions as "wow," "tweeter," "motorboating," "stereo," and all the rest, which provide accurate descriptive names for what we are dealing with and so make them *realer*, as well as easier to correct or use, and more specifically identifiable.

The paramount value of a distinctive audio terminology is obvious in the field of disc recording, where in the course of years a convenient working vocabulary has been devised — and generally accepted — to identify both the specialist collector and the general science of cataloguing or surveying the recorded repertory. Yet, although tremendous technical progress is currently being made in the newer but in many ways closely parallel field of tape recording, all of us who work or play in it, and the mushrooming catalogue of program materials, still suffer from the curse of namelessness.

I've long been vaguely conscious of this intolerable handicap, but it is an inquiry from Philip L. Miller of the Music Division of the New York Public Library which has reminded me that this terminological problem is rapidly becoming a pressing one. And since I can't solve it myself, I'm hoping that the readers of AUDIOCRAFT may have some suggestions. Certainly many of them — no less than librarians and tape reviewers — must have puzzled over the failure so far of the tape industry and recorded-tape fans to provide some appropriate equivalents (*mutatis mutandis*) for the phonographically indispensable terms *discophile* and *discography*.

For my own part, I don't object at all to the homely label of "tape worm," but I must concede that it's much too informal, if not actually pejorative, for serious usage, and that "tapophile" or "tapeophile" would be a hopelessly awkward makeshift. "Tapography," too, seems clumsy and "tapestry," overly facetious or whimsical, yet rack my brains as I will, I haven't yet been able to discover or invent anything better. Perhaps magnetism might serve as a more suitable base for some new coinage, but here again I find myself stymied, although I'm definitely "attracted" to the notion of dubbing tape "disc"ophiles magnetists — i.e. (in the dictionary definition) those skilled in or practitioners of the art of magnetism!



WELL, so far at least, the still unbaptized recorded-tape fan may disdainfully mutter, "What's in a name?" or, "A rose by any other name would smell as sweet." Certainly the lack of one hasn't notably checked the listening public's acceptance of the new medium. But when we remember the usefulness of *discophile* and *discography*, to say nothing of *audiophile* and *high fidelity*, the vocabulary gaps where tape is concerned become painfully evident and it's easier to realize that not only a few library cataloguers, critics, and commentators are at a serious loss, but that everyone concerned in any way with recorded tapes lacks essential expressive tools.

Sooner or later someone is sure to come up with terminology that is either appropriate or catchy enough to take hold. Who'll be the semantic pioneer?

— R.D.D.

fill'er up—WITH MUSIC

by Joseph Rebholz

All photos by the author.

If you turn an auto radio up loud enough to hear it clearly, conversation in a normal tone of voice becomes nearly impossible. Turn the radio down far enough to talk, and the music is lost. Back-seat passengers are another problem; if the radio volume is increased enough for them to hear it well, the little speaker rattling away under the dash panel becomes unbearable to those in the front seat.

Fortunately, there is an inexpensive solution to such problems. Don't try to outguess us, though; simply adding a single rear-seat speaker doesn't do the job.

You see, adding another spot of sound (such as a single speaker) won't help much. Passengers in today's (or any other day's) automobiles are assailed by a continuing racket composed of wind whistling around and through windows, rattling doors on the sides, road noises and seat rattles from below, and engine vibrations and wheezes coming through the firewall. There is also the slap of tires on tar strips, the rumble of a noisy exhaust, and transmission gears whirring. What chance does music have?

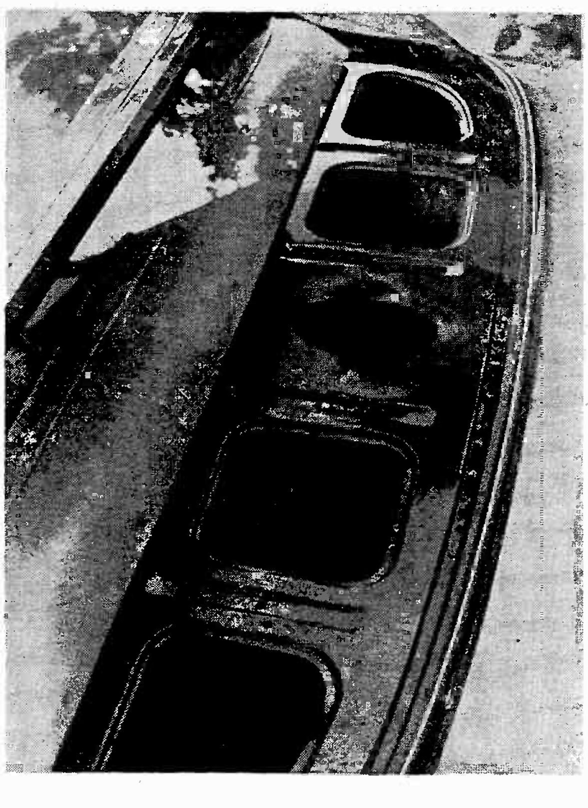
Adjusting a spot source of music, such as the dash-mounted radio speaker, is usually done to satisfy the listening requirements of the ear nearest the sound source (radio speaker). This is fine as long as sound reaching the other ear is of lower intensity. Your brain recognizes only the music, which overrides other miscellaneous sounds. But let any other noise intrude at the same or greater level as the radio, and your brain begins trying to sort out and listen to two different sounds at once. The result: irritation and unhappy confusion. Trying to listen to both noise and music at the same time is almost as tiring as driving itself.

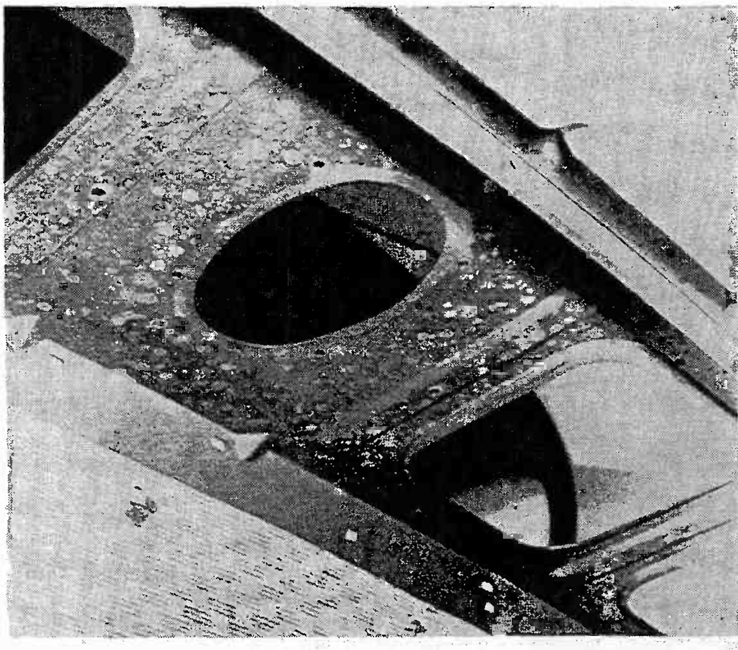
The easiest way to cope with these problems is to wear thick earmuffs. Or, if you are a music lover, turn the body of your car into a gigantic speaker chamber. Fill it so full of music, so evenly spread about, that the gentlest music played at a low level will completely surround and obscure bothersome metal or airborne noises. In this chamber both ears will hear music at the same level, and any outside disturbing noises must be loud indeed to drown out the pleasant melody surrounding you and your passengers.

Installation of *several* extension speakers is the answer. Two or more in the back, and an extra speaker or two in the front, will do a fine job of creating the sound chamber. The package shelf

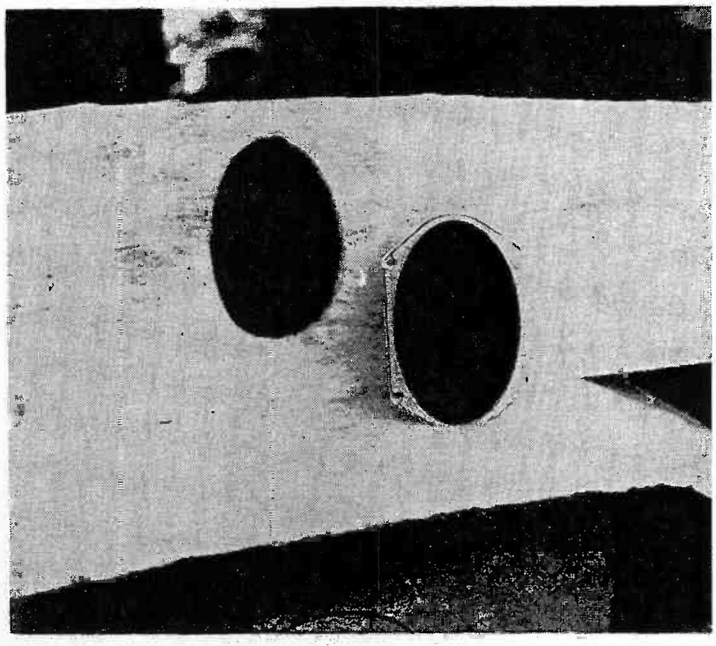
Typical of the equipment available for automobile extension speakers are these 6-, 5-, and 3-inch round speakers, and the standard 5-by-7-inch oval unit. Grilles come in many sizes.

Many cars have several holes in rear shelf that may be suitable for mounting speakers; others have only one hole in the center. Sheet-metal shop can cut new ones.





Bottom view of metal shelf. Fiberglas is sandwiched between this shelf and the Masonite speaker mounting board. Work is easier if trunk lining (held by bent-over clips) is removed.



Masonite or other hardboard is trimmed to fit below package shelf, and speaker holes cut out. Fiberglas batting of same shape is then glued to upper side of speaker mounting board.

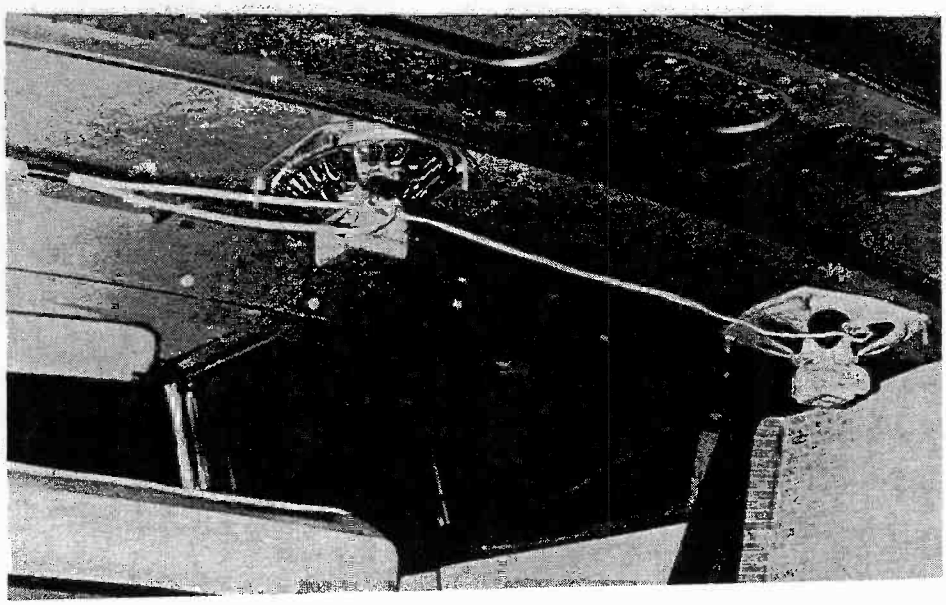
behind the rear seat is usually pre-punched for one extension speaker, but holes are easily added for others. On some makes of car the package shelf is full of holes for air-conditioning equipment. These same holes will do quite nicely for the installation of spare speakers. In front, good locations for added speakers are the kick panels ahead of the front doors near the floor boards. Perhaps a speaker or two under the front seat or dash panel would help in dispersing the sound. Small speakers could be hidden behind the headlining above or behind the sun visors, but it's a good idea to measure the depth of this space *before* buying the speakers.

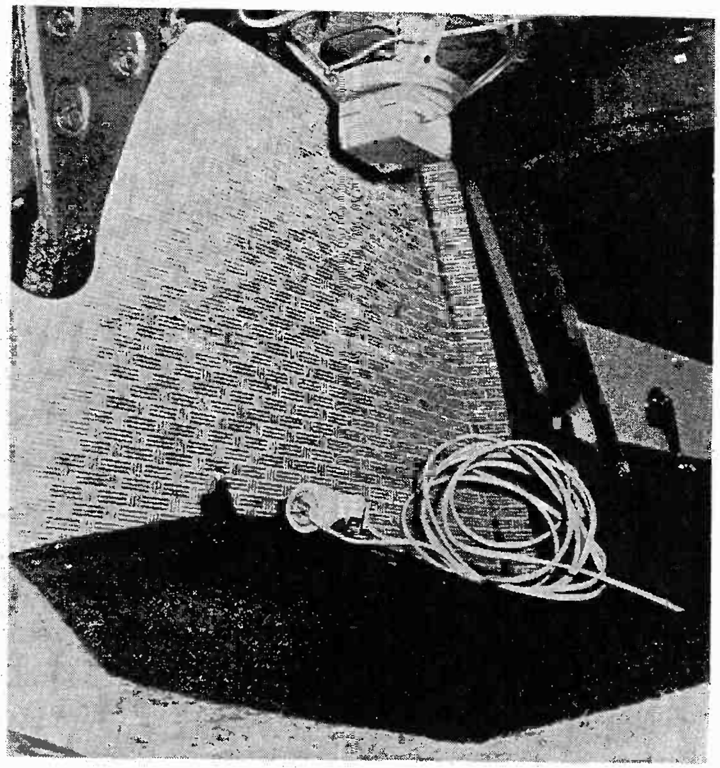
The installation of a number of speakers is advantageous for several reasons. First, those speakers mounted on the rear package shelf are very well baffled by the closed luggage compartment. Because of the trunk's large size and sound-absorbent lining (floor mat and side panels), it makes a nearly perfect infinite baffle. Such a baffle increases the bass response to a considerable extent, and reduces the tinny sound associated with small speakers. Second, using several extension speakers reduces the amount of energy each speaker must handle, to the point where distortion becomes negligible. Third, both large and small speakers can be used, and the small speakers can handle high-frequency sounds better than large units. Fourth, smoother response is obtained from a group of dissimilar speakers than from one, because their resonances average out. Finally, several sound sources give more uniform distribution of sound throughout the car. The result is an illusion of sound coming from every direction, both front and back, which is just what we need to override common body and road noises.

Tools required to install extension speakers are few in number: an electric-drill motor and drills, to make mounting holes for the speakers and grilles; a soldering iron and *rosin*-core solder; a knife to trim the wire and cut holes in cardboard; a keyhole saw, scissors, and a wrench or two to undo the bolts holding the back seat in place; and a screw driver. A wench or two with cool beer helps also.

Before you buy anything, remove the cushion and back of the rear seat. The cushion just lifts out, while the seat back is usually held down by two bolts or clips hidden at the bottom of the seat frame. After these bolts are removed, lift the back straight up (don't try to pull it outward) off the metal hooks which go through the back near the top. Only two or three inches of lift is needed to get the back off its hooks; then it can be pulled outward and taken from the car. Often there is a section of cardboard between the seat back and trunk compartment, which is held in place by bent-over metal clips or snap buttons. The clips can be pried up, or the buttons levered out, with a

Bolts go through metal shelf, Fiberglas, hardboard, and speakers to hold the assembly securely together. If grilles above have mounting holes corresponding to those of the speakers, bolts go through them and cardboard shelf cover too. Check text of article.





Phasing the speakers is necessary to make certain that they work in synchronism. It is easily done with a flashlight cell and color-coded hookup wire, as author explains in the text.

wide-blade screw driver. Take out the cardboard divider gently so it will not be torn. Then carefully remove the cardboard covering of the package shelf. Some of those are glued in, others have buttons, and some just lie in place. Go slowly, for you must work with this part several times before putting it back when the job is completed.

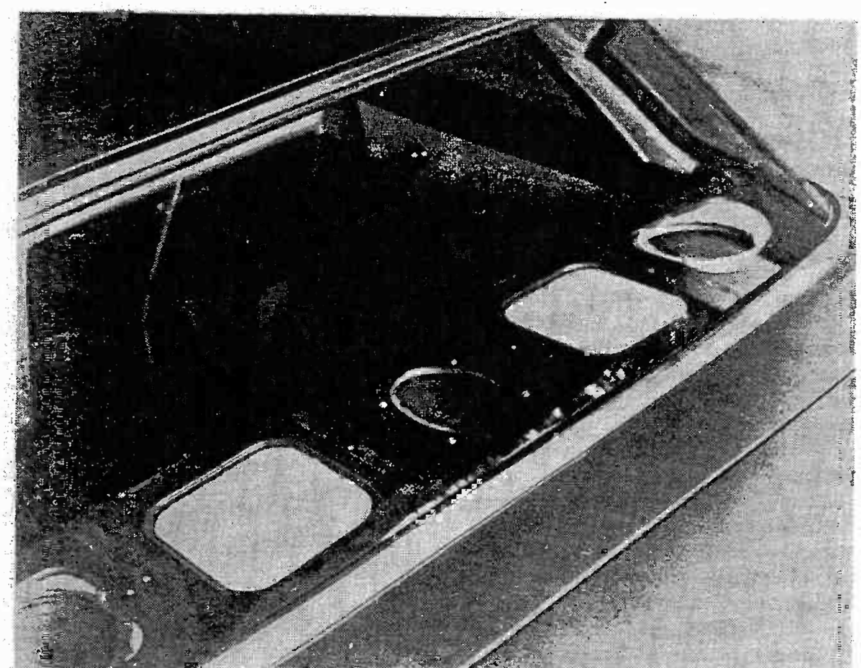
Now the metal base of the package shelf should be exposed. If it has a number of holes (like the shelf pictured with this story), just buy two, three, four, or five speakers that will fit the holes *and* fit under available speaker grilles. Grilles do not come in every size, so it's worth while to buy both speakers and grilles at the same place. Another point to check is that enough room will be available for the grilles after the seat back is reinstalled. Usually, an inch or two of the package shelf is covered by the seat back.

There may be no holes in your car's shelf, or only one for a small single rear speaker. This isn't much of a problem, but will be a bit more expensive. Buy a speaker and grille to match the hole already there, then buy a number of round speakers to mount across the rest of the shelf. Measure each newly purchased round speaker across the inside of the mounting ring (the flange of the speaker with a fiber insert and mounting holes), and have a sheet-metal shop cut holes in the shelf slightly smaller than the mounting-ring inside dimension.

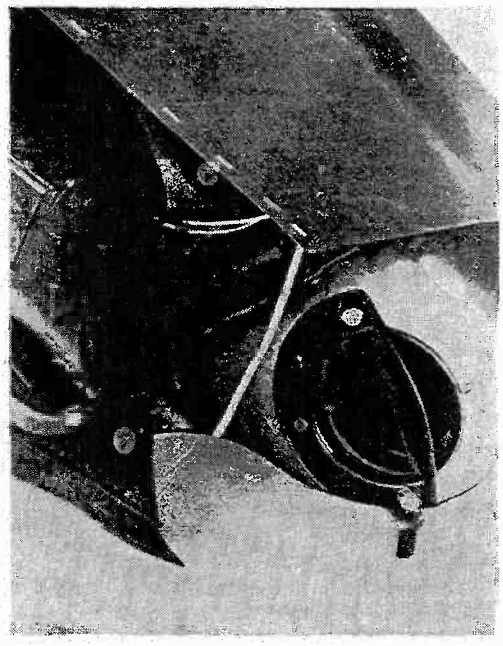
The number of speakers to be used depends on how complete a job you wish. One in each corner is fair, one in each corner with one in the middle is good, and four or five equally spaced across the back will do a really bang-up job. But suit yourself and the pocketbook. At the nearest radio supply house you should be able to obtain everything to do the job. Or, if the postman is easier, check mail-order listings in this magazine for the supplies needed. Prices of speakers will vary, but $2.50 each is about right to pay for 5- or 6-inch round, or 5-by-7-inch oval speakers of fair quality. More expensive speakers will sound better, perhaps, but don't go overboard on high-fidelity units. This installation doesn't merit that kind of quality, and your money will have been poorly spent unless you intend to use a high-quality amplifier too.

Speaker hookup wire can be had for 2¢ per ft., with 25 ft. adequate to do a three-speaker installation. Control-switch kits are useful when attaching the extension speakers to your radio. A dollar buys a simple three-position switch that will let you play the front speakers alone, the rear speakers alone, or the whole outfit together. Another control, which costs around $2.00, permits fading from front to rear and back again, or equalizing the sound between the speakers. This is preferable for a multiple-speaker installation because with it you can balance the units to perfection. It's easy to mark the optimal control-knob setting so that you can return to it quickly after someone else experiments with it.

Unpainted or crackle-finish grilles are suggested (at about 75¢ each), because on the way home you can visit the auto-supply store for a small bottle of auto touch-up paint. Select a matching (or contrasting) color for your car so the painted grilles will have all the snap of an expensive factory installation.



Three speakers installed on the package shelf, ready for the first test. The glass-wool batting deadens panel resonances and makes inexpensive speakers sound almost like hi-fi units.



Extension speaker cable is run forward with light wires. Emerging from behind kick panel, it is draped over the glove compartment and led to switch or radio.

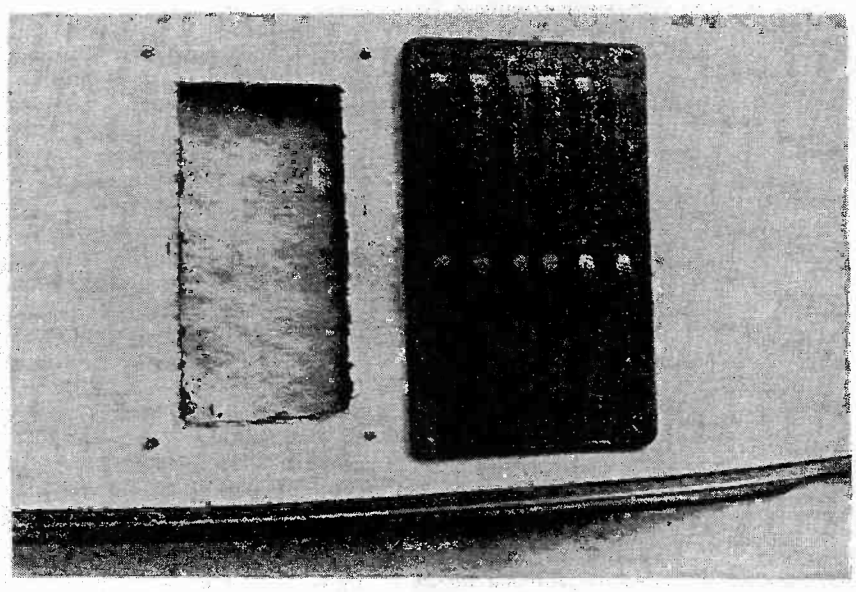
While you're there, buy a package of under-the-hood Fiberglas and a dozen ½-inch sheet-metal screws to hold the grilles down. Of course, you may have been lucky and found grilles that have bolt holes lining up with the speaker mounting holes. In that case omit the sheet-metal screws. Last stop is the hardware store or lumber yard for a piece of ⅛-inch or ¼-inch Masonite or other hardboard, 12 in. wide by 60 in. long. Now you may go home.

First step in the installation is to paint the speaker grilles with the auto touch-up paint. While they're drying, uncover the cardboard package-shelf cover and lay it on the hardboard. Using the shelf cover as a pattern, mark the hardboard and cut it to the same shape as the package shelf, but at least an inch smaller all the way around. The trimmed Masonite is placed under the metal base of the package shelf and goes in from the trunk side. You may have trouble getting it in place, and if so, just cut the Masonite in half across the narrow dimension. Fit each half separately. After the Masonite is fitted snugly up under the metal shelf, mark on it the outlines of all holes in the shelf. Remove the Masonite to a workbench or box and mark the location of each speaker in the center of the package-shelf holes. The speakers must bolt to the hardboard, and only bolt through the metal shelf if it happens to be in the way. Cut out the speaker holes from the Masonite, and use a file or sandpaper on rough edges to eliminate the debris. Now take a single layer of the Fiberglas and glue it to the Masonite, on the side that goes against the metal shelf.

Almost any old pair of scissors will cut the Fiberglas as you trim the edges and around the speaker holes. Put the hardboard panel(s) up under the metal shelf base with the Fiberglas between the metal and Masonite. Drill a couple of holes and bolt the panel in place to hold it up while you install the speaker and grilles.

Should you have been lucky enough to get grilles to match the speakers, proceed as follows. Install the cardboard package shelf in its proper location. Mark through from the bottom of the Masonite to the underside of the cardboard where the grille holes belong in the package shelf. Remove the cardboard shelf and start cutting out the holes — after you have doubly checked to be sure the grilles will fully cover the holes you intend to carve out. Grilles O.K.? Then cut. Reinstall the package shelf, which now gives you speaker cutouts in Masonite, Fiberglas, metal shelf base, and cardboard package shelf — and all should line up in acceptable fashion.

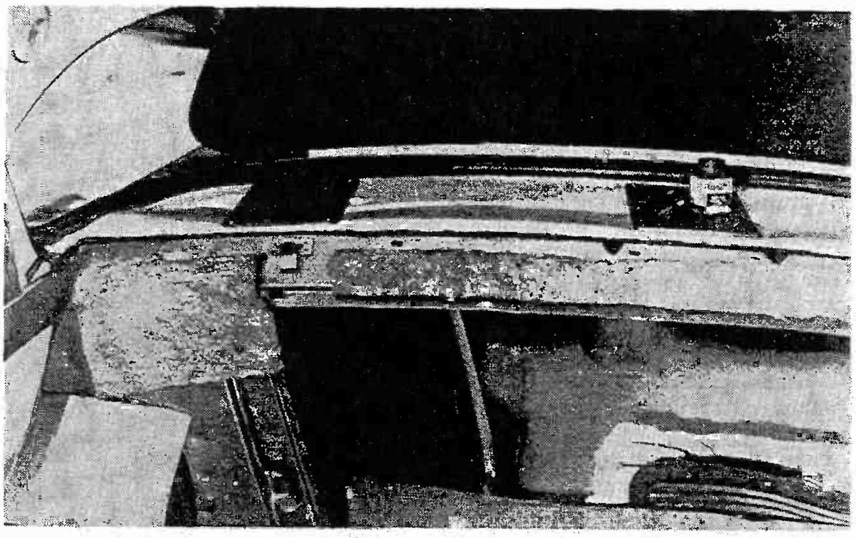
Car radios often have extension speaker plugs and sockets, such as this one. If yours does not, the wires will have to be attached to dash-panel speaker lugs.



Before installing speakers, you've marked the mounting-hole locations on lower side of the cardboard package-shelf cover. Now you can cut grille holes with a knife, and mount grilles.

Hold each speaker up under the Masonite with the soldering lugs toward the rear of the car (they're easier to solder this way) and pencil in the mounting-hole locations. Drill through the pencil marks and pray you drilled so straight that the grille matches the same holes. Use regular nuts and round or button-head bolts to snug up the whole assembly of speaker to grille. By the way, try to get all four mounting bolts in; this reduces the chances for speaker-frame vibrations.

If the grilles and speaker mounting holes do not match up, here's the other way to put your rear-deck music system together. Install the cardboard package shelf and mark the locations of the speakers on the bottom of the cardboard by drawing through the speaker holes in



After touch-up painting the grille mounting screws, the back seat rest can be reinstalled. It drops in place over two or more metal hooks; one can be seen just below grille at left.

the Masonite and metal shelf. Remove the cardboard package shelf and stack it well out of your way. Hold each speaker up against the bottom of the Masonite with the solder lugs toward the rear of the car, and pencil-mark the mounting holes. Drop the speaker out of the way and drill the marked holes. Bolt the speaker in place. The bolts, in this instance, head against the metal shelf (if the shelf hole isn't larger than the speaker hole), go through Fiberglas, Masonite, and finally the speaker mounting ring. Install the remaining speakers in the same manner.

Now, back to the cardboard package shelf. Center a grille over the previously marked speaker hole, keeping the grille at least 1½ in. away from any edge of the cardboard. Mark around the grille and cut a hole through the cardboard about ½ in. inside the grille outline. Now you can be certain the grille will fully cover the cutout. Do all the grille holes this way, clean up the rough edges, and reinstall the package shelf. The speakers and grilles pictured with this article were done in this manner. You may notice that one of the grilles is smaller than the speaker it covers. Nothing to worry about, though; it still sounds wonderful. With the package shelf now back in place, all you have to do is line up the grilles, drill through the grille mounting holes into the metal shelf base, then use the ½-inch sheet metal screws to hold the grille down. In this installation the grille mounting screws pass through the cardboard package shelf and hold both it and grille to the metal shelf base — a really snug assembly.

Use a little of the touch-up paint left from painting the grilles to color all exposed screw heads, so they will match the balance of the installation.

Wiring speakers properly requires a little care. Split each end of the speaker wire for several inches and trim the insulation back on each wire about ¼ in. One of the two wires will be coded by having a tiny strand of colored string running either with the wire or buried in the insulation. This marked wire will be called the coded wire for purposes of identification in these instructions.

Borrow a single flashlight cell and solder one end of the coded wire to the top center button of the dry cell. Solder the uncoded wire to the bottom of the battery. Then take the other ends of the wire and touch one to each lug of a speaker. The speaker cone will jump either inward or outward. Try the wires both ways to make certain the speaker cone jumps outward from current in the dry cell. Then mark the speaker lug to which you were holding the coded wire, when the cone jumped outward, with a "P" for Positive. This is known as "phasing" the speakers; it is very important, because the cones must all move in the same direction at the same time. Check them all in the same manner, switching wires until the cone jumps outward, marking the coded lug with a "P." After they are all marked, clip off the battery and sneak it back into the household flashlight.

Now connect the speakers by tying all the P lugs together, using the coded side of the two-section speaker wire. Use the unmarked or uncoded strand of wire to connect the unmarked speaker lugs together. It may be easier for you to run short lengths of cut wire between the speakers and then connect the long wire to that speaker nearest the car wiring. After all connections are soldered tightly, run your main wire with car wiring going from the tail light or trunk light. String the new wire in the same clips and tape it as seems necessary to keep it from fraying or rubbing on sharp metal edges. If you have trouble finding the auto wiring, look under the floor mat near the driveshaft tunnel, or near the door sills. It's usually in one place or the other. Checking the speakers is done by holding the ends of the main wire against lugs on the dash-panel speaker while the radio is operating. If your new speakers come to life, everything is O.K. so far. Now's a good time to

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The final pleasant task is adjustment of the relative volume levels between front- and rear-mounted speakers. Fading controls for this purpose can be obtained at radio parts stores.

by **HAROLD REED**

Free Energy for Transistors

SURPRISING as it may seem, you can build and experiment with transistor circuits that require no batteries or AC power supplies. Of course, energy from some source is necessary to obtain the required transistor action. If we do not supply this energy, then we must obtain it for our transistor apparatus from natural or man-made sources. Power from these sources can be termed "free power," because it costs the user nothing once the component parts are obtained and assembled.

Since a transistor in a suitable circuit can be operated with a minute amount of energy, this free energy may be obtained from the sun by means of photoelectric cells. Many circuits using this method have been described. Operating power can also be obtained from sound energy, or from the electromagnetic field radiated by a radio transmitter. I have investigated these different power sources, and have found the electromagnetic radiation source to be the most intriguing and reliable. Two circuits for recovering this energy from radio waves are discussed in this article. I have used them to power several transistor circuits, two of which are shown connected to the free-power circuits.

Fig. 1A is a simple circuit that can be used to obtain rectified current from a signal radiated by a local AM broadcasting station. Here a loopstick inductor and 365-$\mu\mu$fd tuning capacitor are tuned to the frequency of a local station. The magnitude of the rectified current will depend on the radiated power of the station, the distance between the station and the receiving location, and the efficiency of the receiving equipment in picking up the radiated signal. RF energy is rectified by a 1N34 or CK705 diode, and the DC current flows through the load resistor, R1. The current varies, of course, with adjustment of potentiometer R1.

A .001-μfd capacitor is used to filter out the RF component of the diode input. The DC component varies in accordance with the station's audio modulation. Fig. 1B is a pictorial wiring diagram showing the recommended tuning components; others can be substituted. If a grounded chassis is not used to build this power supply, the foot lugs of the terminal strip should be connected to a ground such as a water or steam pipe.

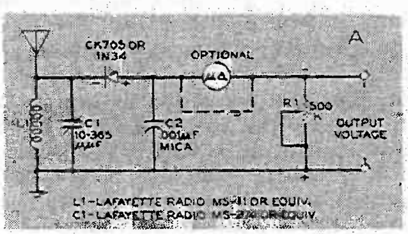


Fig. 1A, above: Very simple AM tuner and detector circuit. Its output is DC varying at audio modulation rate.

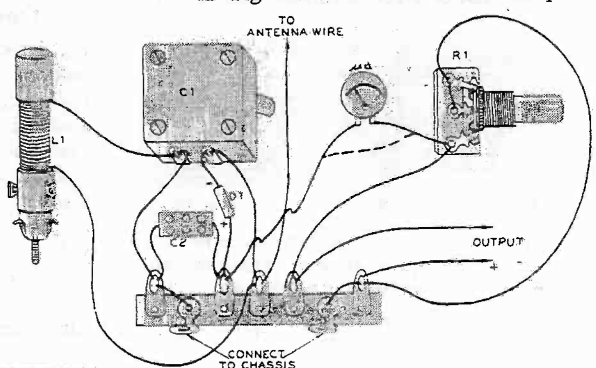
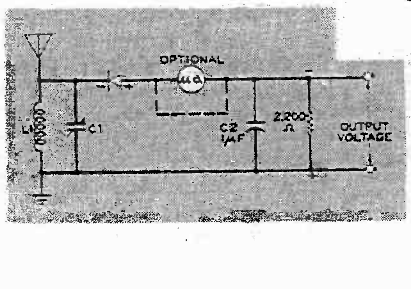


Fig. 1B, right: Circuit above shown in a pictorial wiring diagram.

Fig. 1C, below: This revised circuit has a filter to remove AC from the DC output terminals.



Using a 30-foot length of No. 14 wire hung in the attic, for an antenna, and a water-pipe ground, current readings up to 200 μa were indicated by the microammeter at this location. The polarities are as shown in Fig. 1; if the diode connections are reversed, the output-voltage polarity will be reversed. With a pair of high-impedance headphones in place of R1, current readings up to 116 μa were obtained and, of course, the audio modulation signal could be heard in the phones. Since for this purpose the audio component was not desired, it was bypassed with a large capacitor across the output, as shown in Fig. 1C. The larger this capacitor, the more effective is the audio suppression. A 1-μfd capacitor did a good job. A 2,200-ohm resistor is shown in Fig. 1C as a load resistor in place of the phones. As long as the AM station continues broadcasting, the voltage developed across the load resistor is available to furnish power for transistor circuits.

It was decided to use this free energy to power a transistor audio oscillator,

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TRANSISTORS *in Audio Circuits*

by PAUL PENFIELD, JR.

VIa: Single-Stage Analysis

IN Part 5 of this series we discussed linear low-frequency small-signal incremental equivalent circuits for transistors. The idea was that the transistor symbol in circuit diagrams could be replaced, for the AC analysis, by this equivalent circuit. Since the circuit is a linear one, the whole diagram may be expressed by algebra alone, without recourse to the characteristic curves.

In this part we will use the equivalent circuit to examine the properties of transistor amplifier stages. Often, the information derived this month can be used by itself to design and analyze transistor amplifiers, eliminating the trouble of using even the equivalent circuit.

Important properties of transistor stages are the voltage gain, the current gain, the input resistance, and the output resistance. Other quantities of occasional interest include the power gain, the transducer gain, and the system transconductance. We will derive formulas for each of these quantities, and indicate

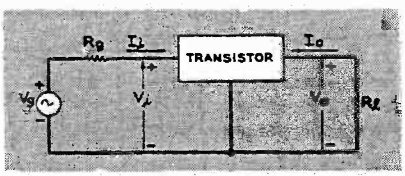


Fig. 1. The single-stage AC quantities.

how to use them. In many cases the formulas are quite simple. First we will find the formulas, and then show how to reduce complicated diagrams into the simple form assumed.

Ideal Transistor Stage

Fig. 1 shows what we call an ideal transistor stage. It consists of a transistor, a source with an internal resistance, and a load resistance. In general, any audio amplifier stage can be reduced *for the purposes of AC signal calculations* to the form of Fig. 1. Later on we will see how this is done.

Since Fig. 1 can be obtained from virtually any transistor stage, it should be obvious that, if we solve for the important quantities completely in Fig. 1, then in a sense we have solved all transistor audio amplifier stages. It will only be necessary to reduce the diagram to the form of Fig. 1, and use the formulas presented below. Usually this method is the easiest to use in designing audio amplifiers, although it will not work in all cases.*

In order to make this derivation as general as possible, we will obtain formulas for each configuration: grounded-emitter, grounded-base, and grounded-collector, although the last two will probably be used only occasionally. Each set of formulas will be given in terms of both the commonly used parameter sets — the *r* set, and the h_b set. To illustrate the method of deriving the formulas, however, we will compute them for the grounded-emitter stage using the grounded-emitter *h*-parameter set.

Deriving the Formulas

Fig. 1 shows the transistor with a general voltage source and a load. The input and output voltages and currents are assumed from now on to be in the directions indicated. The *current gain* A_i is defined as the ratio of output current to input current, I_o/I_i. The *voltage gain* A_v is defined as the ratio of the output voltage to the input voltage, or V_o/V_i.

The *input resistance* R_i is defined as the ratio of the input voltage to the input current, or V_i/I_i. This is the resistance that the external circuit to the left of the transistor "sees" as the transistor.

The *output resistance* R_o likewise is defined to be the resistance that the circuit to the right of the transistor "sees". However, it is *not* the ratio of output voltage to output current, for this value gives the load resistance R_l (or, rather, the negative of it). Instead, it is most easily calculated as the ratio of output voltage when the output is open-circuited, to the output current when the output terminals are shorted. Thus,

$$R_o = -\frac{V_o \text{ (when } R_l = \infty)}{I_o \text{ (when } R_l = 0)} \quad \ldots(1)$$

Now the less important relationships: the *system transconductance* g_s is defined as the ratio of output current to source voltage, or I_o/V_g. This is useful, generally, only with input stages.

The *power gain* G is defined as the ratio of power delivered to the load, to the power delivered to the transistor input terminals. Thus, it is $-V_oI_o/V_iI_i$, which is simply the product of $-A_v$ and A_i. The *transducer gain* G_t is defined as the ratio of power delivered to the load, to the maximum possible power obtainable from the source. Maximum source power is $V_g^2/4R_g$, and so

$$G_t = \frac{V_oI_o4R_g}{V_g^2} = 4R_lR_gg_s^2 \quad \ldots(2)$$

Thus, in summary, the quantities we wish to determine are as follows:

$$R_i = \frac{V_i}{I_i} \quad \ldots(3)$$

$$R_o = -\frac{V_o(R_l = \infty)}{I_o(R_l = 0)} \quad \ldots(4)$$

$$A_v = \frac{V_o}{V_i} \quad \ldots(5)$$

$$A_i = \frac{I_o}{I_i} \quad \ldots(6)$$

$$g_s = \frac{I_o}{V_g} \quad \ldots(7)$$

$$G = -A_vA_i \quad \ldots(8)$$

$$G_t = 4R_gR_lg_s^2 \quad \ldots(9)$$

If we now specialize in the grounded-emitter case, and use the *h*-parameter equivalent circuit, we obtain the dia-

Text continued on page 43

*Sometimes it is not possible to reduce the circuit to the form of Fig. 1 — when single-stage feedback is used, for example. Also, if there are frequency-compensating networks in the stage, it is best to go back to the equivalent circuit, rather than to try to use this analysis.

Table I

$$R_{ie}=r_b+r_e+\beta r_e\left(\frac{r_d}{R_l+r_d}\right)=\frac{h_{ib}+R_l(h_{ib}h_{ob}-h_{rb}h_{fb})}{1+h_{fb}+R_lh_{ob}}$$

$$R_{oe}=r_d+\beta r_d\left(\frac{r_e}{R_g+r_b+r_e}\right)=\frac{h_{ib}+R_g(1+h_{fb})}{R_gh_{ob}+h_{ib}h_{ob}-h_{fb}h_{rb}}$$

$$A_{ve}=\frac{-\beta R_l}{\beta r_e+(r_b+r_e)\left(1+\frac{R_l}{r_d}\right)}=\frac{h_{fb}R_l}{h_{ib}+R_l(h_{ib}h_{ob}-h_{fb}h_{rb})}$$

$$A_{ie}=\frac{\beta r_d}{r_d+R_l}=\frac{-h_{fb}}{1+h_{fb}+R_lh_{ob}}$$

$$g_{se}=\frac{\beta r_d}{(r_d+R_l)(r_b+r_e+R_g)+\beta r_er_d}$$

$$=\frac{-h_{fb}}{h_{ib}+R_l(h_{ib}h_{ob}-h_{fb}h_{rb})+R_g(1+h_{fb}+R_lh_{ob})}$$

$$G_e=-A_{ve}A_{ie}$$

$$G_{te}=4R_gR_lg_{se}^2$$

Table II

$$R_{ib}=r_e+r_b\left(\frac{R_l+r_d}{R_l+r_c}\right)=h_{ib}-\frac{h_{fb}h_{rb}R_l}{1+R_lh_{ob}}$$

$$R_{ob}=r_c\left(1-\frac{\alpha r_b}{R_g+r_e+r_b}\right)=\frac{h_{ib}+R_g}{R_gh_{ob}+h_{ib}h_{ob}-h_{fb}h_{rb}}$$

$$A_{vb}=\frac{\alpha R_l}{r_e+r_b(1-\alpha)+(r_e+r_b)\left(\frac{R_l}{r_c}\right)}$$

$$=\frac{-h_{fb}R_l}{h_{ib}+R_l(h_{ib}h_{ob}-h_{fb}h_{rb})}$$

$$A_{ib}=\frac{-\alpha r_c}{r_c+R_l}=\frac{h_{fb}}{1+R_lh_{ob}}$$

$$g_{sb}=\frac{-\alpha r_c}{(r_e+R_g)(r_c+R_l)+r_b(R_g+r_d)}$$

$$=\frac{h_{fb}}{(R_g+h_{ib})(1+R_lh_{ob})-R_lh_{fb}h_{rb}}$$

$$G_b=-A_{vb}A_{ib}$$

$$G_{tb}=4R_lR_gg_{sb}^2$$

Table III

$$R_{ic}=r_b+r_e(1+\beta)+\frac{r_cR_l}{r_d+R_l}=\frac{h_{ib}+R_l}{R_lh_{ob}+1+h_{fb}}$$

$$R_{oc}=r_e+r_b(1-\alpha)+\frac{r_dR_g}{r_c+R_g}=\frac{h_{ib}+R_g(1+h_{fb})}{1+R_gh_{ob}}$$

$$A_{vc}=\frac{R_l}{R_l+r_e+r_b(1-\alpha)}=\frac{R_l}{h_{ib}+R_l}$$

$$A_{ic}=\frac{-r_c}{R_l+r_d}=\frac{-1}{R_lh_{ob}+(1+h_{fb})}$$

$$g_{sc}=\frac{-1}{\left(\frac{R_g+r_c}{r_c}\right)R_l+r_e+(1-\alpha)(R_g+r_b)}$$

$$=\frac{-1}{h_{ib}+R_l+R_g(1+h_{fb}+R_lh_{ob})}$$

$$G_c=-A_{vc}A_{ic}$$

$$G_{tc}=4R_lR_gg_{sc}^2$$

Table IV

Assuming $R_l<<r_d$ *or* $R_l<<\frac{1+h_{fb}}{h_{ob}}$ *and*

$R_g>>r_b+r_e(1+\beta)$ *or* $R_g>>\frac{h_{ib}}{1+h_{fb}}$:

$$R_{ie}=r_b+r_e(1+\beta)=\frac{h_{ib}}{1+h_{fb}}$$

$$R_{oe}=r_d=\frac{1+h_{fb}}{h_{ob}}$$

$$A_{ve}=\frac{-\beta R_l}{r_b+r_e(1+\beta)}=\frac{h_{fb}R_l}{h_{ib}}$$

$$A_{ie}=\beta=\frac{-h_{fb}}{1+h_{fb}}$$

$$g_{se}=\frac{\beta}{R_g}=\frac{-h_{fb}}{R_g(1+h_{fb})}$$

$$G_e=\frac{\beta^2R_l}{r_b+r_e(1+\beta)}=\frac{(h_{fb})^2R_l}{h_{ib}(1+h_{fb})}$$

$$G_{te}=4\beta^2\left(\frac{R_l}{R_g}\right)=4\left(\frac{h_{fb}}{1+h_{fb}}\right)^2\left(\frac{R_l}{R_g}\right)$$

Table V

Assuming $R_l<<r_c\left(1-\frac{\alpha r_b}{r_b+r_e}\right)$ *or* $R_l<<\frac{h_{ib}}{h_{ib}h_{ob}-h_{fb}h_{rb}}$

and $R_g>>r_b+r_e$ *or* $R_g>>\frac{h_{ib}h_{ob}-h_{fb}h_{rb}}{h_{ob}}$:

$$R_{ib}=r_e+r_b(1-\alpha)=h_{ib}$$

$$R_{ob}=r_c=\frac{1}{h_{ob}}$$

$$A_{vb}=\frac{\alpha R_l}{r_e+r_b(1-\alpha)}=\frac{-h_{fb}R_l}{h_{ib}}$$

$$A_{ib}=-\alpha=h_{fb}$$

$$g_{sb}=-\frac{\alpha}{R_g}=\frac{h_{fb}}{R_g}$$

$$G_b=\frac{\alpha^2R_l}{r_e+r_b(1-\alpha)}=\frac{(h_{fb})^2R_l}{h_{ib}}$$

$$G_{tb}=4\alpha^2\left(\frac{R_l}{R_g}\right)=4(h_{fb})^2\left(\frac{R_l}{R_g}\right)$$

Table VI

Assuming $r_e+r_b(1-\alpha)<<R_l<<r_d$

or $h_{ib}<<R_l<<\frac{1+h_{fb}}{h_{ob}}$

and $r_c>>R_g>>r_b+r_e(1+\beta)$ *or* $\frac{1}{h_{ob}}>>R_g>>\frac{h_{ib}}{1+h_{fb}}$:

$$R_{ic}=R_l(1+\beta)=\frac{R_l}{1+h_{fb}}$$

$$R_{oc}=R_g(1-\alpha)=R_g(1+h_{fb})$$

$$A_{vc}=1=1$$

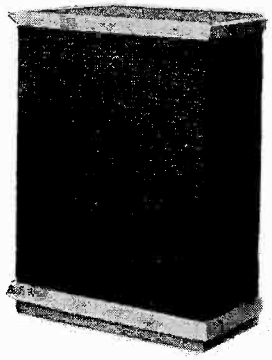
$$A_{ic}=-(1+\beta)=\frac{-1}{1+h_{fb}}$$

$$g_{sc}=\frac{-1}{R_l+R_g(1-\alpha)}=\frac{-1}{R_l+R_g(1+h_{fb})}$$

$$G_c\approx1+\beta=\frac{1}{1+h_{fb}}$$

$$G_{tc}=\frac{4R_lR_g}{[R_l+R_g(1-\alpha)]^2}=\frac{4R_lR_g}{[R_l+R_g(1+h_{fb})]^2}$$

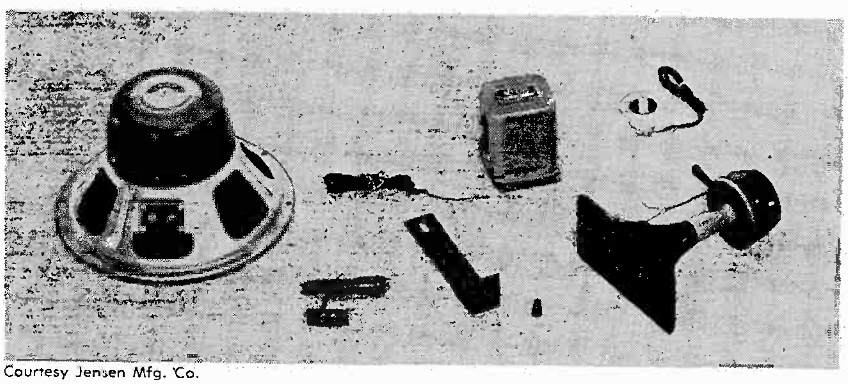
An AUDIOCRAFT *kit report*



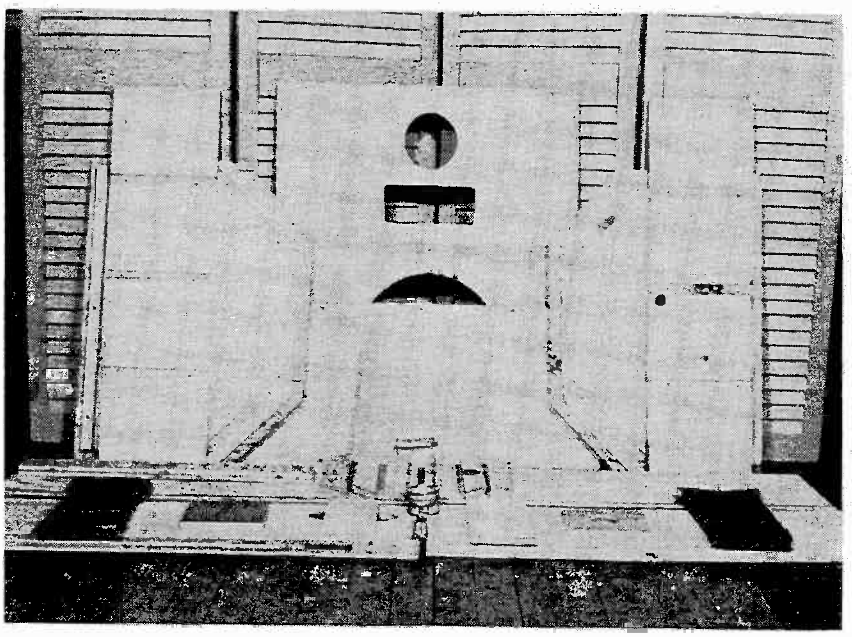
Photos by Warren Syer

Jensen-Cabinart Speaker System

JENSEN Manufacturing Company and the Cabinart Division of G & H Wood Products Company have evolved a unique collaborative method of marketing speaker-system kits. Speaker components for all famous Jensen ready-made speaker systems (Duette, Concerto 12, Concerto 15, Tri-Plex, and Imperial) are available as kits from Jensen. Cabinart makes a complete line of Jensen-designed enclosures in kit form for these driver components. Thus the speaker kits and enclosure kits are designed specifically for each other —



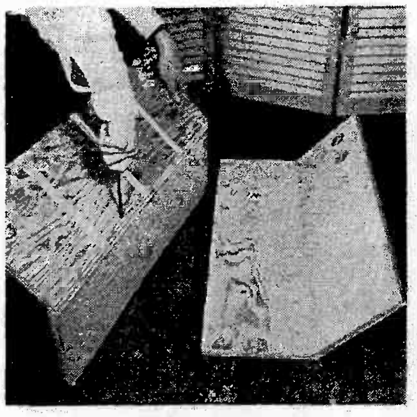
Courtesy Jensen Mfg. Co.

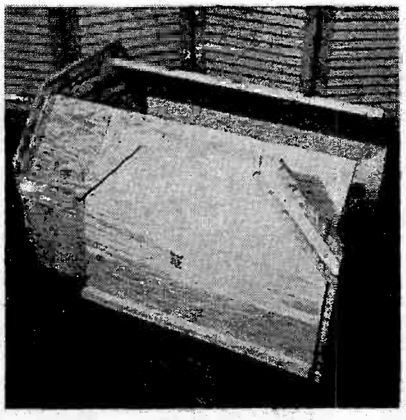


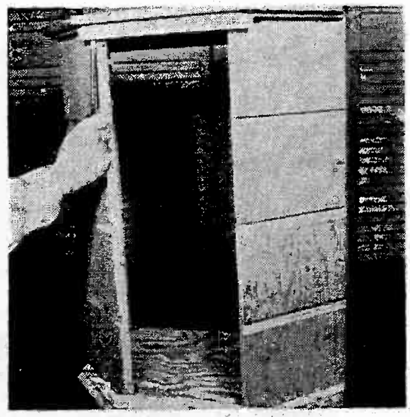
Jensen Concerto 12 speaker-components kit, above; Cabinart's K-107 12-inch Bass Ultraflex enclosure kit, at left.

which assures optimal performance from each — but are individually manufactured by the firms which can make them most skillfully and efficiently.

The Jensen speaker-system kits include the KT-31, for the three-way horn-loaded Imperial; the KT-32 Tri-Plex, for three-way Bass Ultraflex systems; the KT-21 Concerto 15, for two-way Bass Ultraflex systems with 15-inch woofer; the KT-22 Concerto 12, for two-way Bass Ultraflex systems with 12-inch woofer; the KTX-1 Range Extender supertweeter kit; the KDU-10, for 2½-cubic-foot Bass Ultraflex systems or the Treasure Chest Duette, with 8-inch woofer and compression-driver tweeter; the KDU-11, for the Table Duette, with 6-by-9-inch woofer and compression-driver tweeter; and the KDU-12, for a







Enclosure assembly begins with interlocking side pieces. Then top and bottom panels are added, and spacer cleats installed.

budget Duette system with 6-by-9-inch woofer and direct-radiator tweeter. Each kit contains all driver components, crossover networks, level controls, wiring harnesses, and mounting hardware as required for the system.

Cabinart enclosure kits for these systems include the K-101 Imperial folded corner horn, for the KT-31; the K-103 (upright) and K-105 (low-boy) Bass Ultraflex enclosures, for two- or three-way 15-inch systems such as the KT-32 or KT-21; the K-107 (upright) and K-109 (low-boy) Bass Ultraflex enclosures, for two-way 12-inch systems such as the KT-22, with cutout for expansion to three-way system; the K-111 upright Bass Ultraflex enclosure, for 8-inch two-way systems such as the KDU-10; and the K-113 Duette enclosure, for 8-inch two-way systems such as the KDU-10. Each kit contains all wood parts, screws, glue, acoustical padding, plastic wood and sandpaper, base glides, cover plates for all driver and level-control cutouts except that for the woofer, and illustrated assembly instructions. They can be used with coaxial or single-cone wide-range speakers rather than the driver-system kits, if desired. For each enclosure kit there is available at extra cost a prefinished dress kit; this consists of grille cloth, a finished single-piece top panel and molding assembly, a finished single-piece base and molding assembly, and legs if required. The basic kit can be painted or veneered to your taste or, with the dress kit, you can convert it to a finished piece of furniture in less than an hour.

From this wide range of systems, very elaborate to very inexpensive, we chose as being representative the KT-22 Concerto 12 speaker system and the K-107 enclosure with P-207 dress kit. The KT-22 consists of a 12-inch woofer (the Jensen P12-NL), the RP-102 tweeter horn and driver, the A-204 2,000-cps crossover network, the ST-901 high-frequency balance control, horn mounting bracket, and color-coded interconnection cables; the kit price is $73.00. Impedance of this system is 16 ohms, and the power rating is 25 watts. The K-107 enclosure kit is of the upright Bass Ultraflex type, about 33 in. high, 24 in. wide, and 17 in. deep. Like all the upright Bass Ultraflex enclosures, this is designed to fit nicely in a corner but can be used against a flat wall with a slight loss of bass. It has cutouts in the speaker mounting panel for a 12-inch woofer, the RP-102 high-frequency horn, and the RP-302 supertweeter. Cover plates are furnished for both horns, so that neither, one, or both may be used. Price of the K-107 is $39.00; the P-207 prefinished dress kit (Korina blond or mahogany) is $36.00. Total price of the system, then, is $112 without the dress kit, or $148 with it. This should be compared with the factory-assembled prices of the Concerto system, which utilizes the same components: $189.50 in mahogany, and $194.50 in Korina blond.

Construction Notes

Everything has been made as easy as possible for the assembler of the K-107. No hole-cutting or drilling is required; the large holes have all been cut, and all screw holes drilled and countersunk. All parts in our kit fitted precisely. The only tool needed was a screw driver.

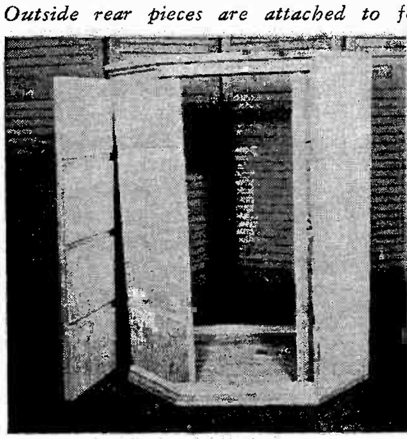
The illustrated instructions are so clear that we can add only three comments which might be useful to the beginner at kit-building:

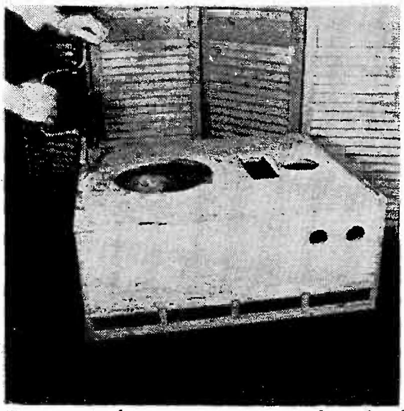
1) Since pieces 3 and 6 interlock along their edges, and pieces 4 and 7 do also, it is easier to assemble 3 to 6 and 4 to 7 before attaching any of them to the top and bottom panels. This assembly procedure is shown in the photographs with this article.

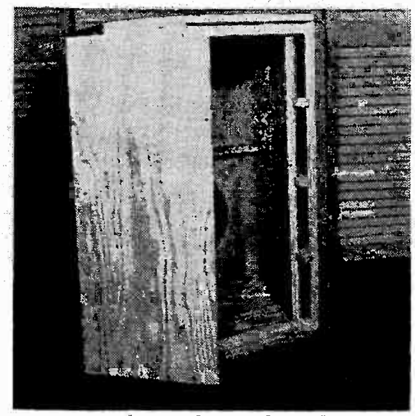
2) Before performing step 3, install the back panel (piece 15) temporarily, centering it on the top and bottom panels. Then, as you screw pieces 9 and 10 into place, push them firmly against piece 15, so that piece 15 will make a good seal when it is put on later.

3) Don't forget to install a cover plate over the unused level-control and supertweeter holes — but don't glue the cover plates, because you may want

Outside rear pieces are attached to form Ultraflex tunnels. Tunnel spacer-stiffeners, front-panel brace are glued in place.







Front panel may or may not be glued; see text. Back panel completes basic enclosure. At right, P-207 prefinished dress kit.

to add a supertweeter later. The speaker components can be attached to the front panel more easily before it is secured to the enclosure. If you don't plan to finish the exterior, access to the speakers is possible (and more convenient) by removing the front panel, so you may want to attach it with screws only, gluing the back panel in place. But if you have the dress kit the front panel can't be removed after the grille cloth is put on, so you may as well glue on the front panel. In that case, don't glue the back panel; simply attach it with screws, and use a nonhardening calking compound to seal it. And don't forget to brush on a fast coat of flat black paint or enamel on all exterior surfaces that will be covered by grille cloth; otherwise, the wood will show through!

It took us just 3½ hours to assemble the enclosure and mount the speakers, after the trial run (without glue) during which the pictures shown here were taken.

Speaker installation will present no problems except to the ultraperfectionist, who will have to drill holes in the front panel for speaker mounting bolts. Bolts are to be preferred, we admit, although we've never had loosening trouble with round-head wood screws if we used lock washers with them. The horn support bracket was a little short, but screws through the horn flange holes in the front panel held it securely enough. The color-coded hookup wires, together with the clear wiring diagram, would make wiring of the system a snap for anyone. You don't even need to do any soldering; all connections are made with screw terminals.

Audiocraft Test Results

The Bass Ultraflex enclosure is a variant of the bass-reflex type. To understand its principle of operation, it is necessary to recall that any speaker has a resonance, at some low frequency, when its springy suspension system tends to keep the weight of the cone vibrating violently in and out — just as a heavy weight suspended on a spring will vibrate up and down when set in motion. At this natural resonant frequency, which varies from speaker to speaker, a small electrical input signal will produce a relatively large sound output if the speaker is mounted on a plain flat baffle or in an open-backed cabinet. There is a response peak at that frequency and, unless the speaker is especially designed for infinite-baffle mounting, the peak is objectionable in itself. The speaker is also inclined to "sound off" at its resonant frequency whenever it is excited by sudden bursts of sound energy; that is, it has poor transient response. It may be overloaded easily at its resonant frequency. Finally, the speaker's response falls off rapidly below resonance.

A bass-reflex enclosure (developed originally by Jensen) is a cabinet completely sealed except for the speaker mounting hole and one other hole, or port, which tunes the cavity resonance of the enclosure to the same frequency as the speaker's natural resonant frequency. When the speaker is mounted in such an enclosure, the enclosure does

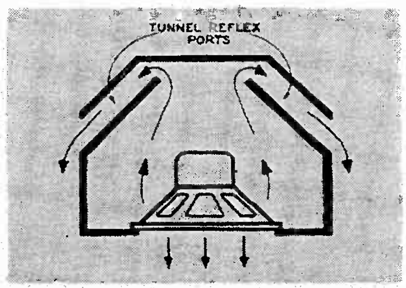


Fig. 1. Bass Ultraflex enclosure design.

three important things if properly designed:

1) It gives the speaker a heavy load to work against at its resonant frequency, thereby reducing markedly the cone travel and distortion, and greatly reducing the possibility of overdriving the speaker.

2) It reverses the polarity or phase of the sound from the back of the cone, which in an infinite baffle is absorbed to prevent cancellation of the front wave, and adds it to the front wave in phase. This increases the efficiency of the speaker in a range of frequencies around the bass resonance.

3) It reduces the response peak at the resonance frequency, and extends the response of the system downward significantly.

The bass reflex, when sturdily built, properly designed, and tuned to the speaker it is used with, gives excellent results with conventional speakers. Two arguments occasionally directed against it are the fact that a fairly close match in enclosure and speaker resonance is required, and that the transient response is not so good as that of a fine bass horn. These arguments have been well answered by recent resistive-loading developments in bass-reflex design. By introducing acoustic resistance in the reflex tuning port, the sharpness of the enclosure-cavity resonance is reduced, or damped. That spreads it out over a greater frequency range, so that matching to the speaker is far less critical; moreover, the transient response is improved, and the response range is extended still further downward.

There are several ways to add the acoustic resistance. It is done simply and effectively in Bass Ultraflex enclosures by substituting thin, long tunnels for the port, as shown in Fig. 1. The friction that air encounters in these tunnels provides the requisite resistance and acoustic damping.

The K-107 is made entirely of ¾-inch plywood; the design is such as to produce an inherently rigid structure; and the only panel that is not braced by other structural members, the front panel, has a stiffening cleat across the middle. The result is a very strong enclosure with no traces of audible panel resonance. It is one of the very few which, in our opinion, are entirely adequate in this respect. This rigidity provides tangible benefits in reduced enclosure coloration and absence of listening fatigue — no mean virtues in themselves.

Although Jensen doesn't publish response specifications for its speakers,

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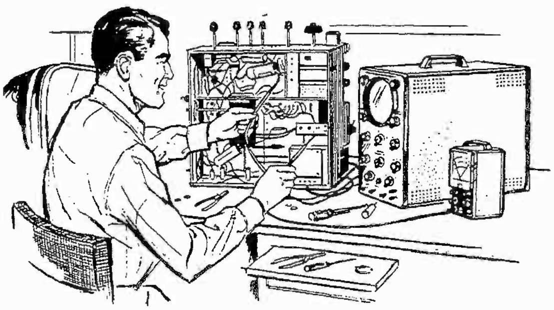


Using Test Instruments

The Oscilloscope, PART II

by DONALD CARL HOEFLER

THE cathode-ray oscilloscope is used primarily for the observation of electrical wave forms, but it is also useful for measurement of such voltages as audio transients, on which the vacuum-tube voltmeter or ordinary multitester is practically useless. Since meter movements have extremely high inertia, voltage peaks of very short duration are gone before the pointer has a chance to move. But the vertical beam deflection in a good modern scope, having excellent high-frequency response, will measure these momentary signals quite accurately, provided it has been properly calibrated. The only accessory required is a plastic grid screen which fits over the face of the tube, as shown in Fig. 1.

We learned in the first article in this series (AUDIOCRAFT, Dec. 1955) that there are several ways to express the value of an AC voltage. Among these is the peak value, which is the absolute maximum value in one direction that the voltage reaches. If we consider both the negative and positive pulses of a given cycle, it is obvious that the *peak-to-peak* figure is just twice the ordinary peak value. This is shown graphically in Fig. 2.

Another expression for AC voltage values, and the one most commonly employed, is the root-mean-square (RMS) quantity. This is the figure which tells the amount of DC voltage that would be required to do the equivalent work of the AC on a resistive load. It is therefore often referred to as the *effective* value. Most ordinary AC voltmeters are calibrated in effective values only, while many vacuum-tube voltmeters read both peak and effective figures.

The average value of an AC voltage is, as its name implies, simply the arithmetic average of all of its instantaneous magnitudes (without regard to direction). Because this value is seldom useful or necessary, it is hardly ever encountered outside of classrooms.

Simple mathematical formulas for these relationships can be expressed as follows:

Effective = peak $\times$ 0.707 = average $\times$ 1.110.

Average = peak $\times$ 0.637 = effective $\times$ 0.901.

Peak = effective $\times$ 1.414 = average $\times$ 1.572.

Since the beam deflection in the scope follows both the negative and positive pulses of the applied voltage, the visual indication on the screen is actually peak to peak. But as the relationship between peak, effective, and average values is constant for a sine wave, the indication on the screen may be taken as representing any one of them.

Calibration of the scope for use as a fast-acting AC voltmeter is then quite simple. All that is required is a source of sine-wave audio, such as an audio-

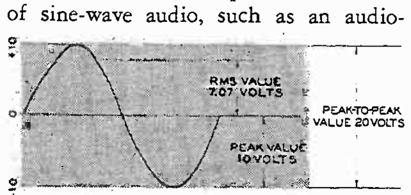


Fig. 2. AC value is expressed in RMS amplitude, but scope shows peak-to-peak.

signal generator, and an AC voltmeter as a reference standard. Since we are not interested in wave forms at this point, but only in vertical deflection as observed on the grid screen, the horizontal selector on the scope is switched to the HORIZONTAL INPUT position. This removes any internal synchronization from

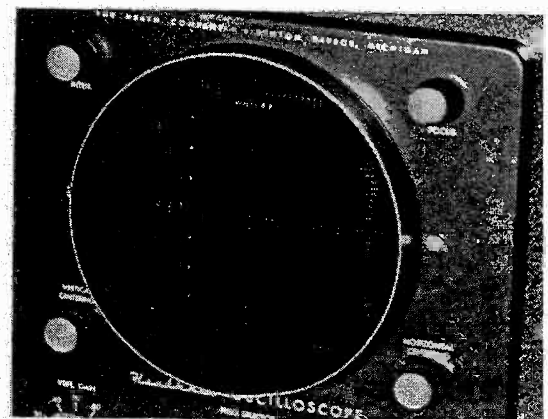


Fig. 1. Calibration mask for a scope.

the horizontal plates, and as long as no voltage is connected to the HORIZONTAL INPUT binding posts, no horizontal deflection will be noted.

The output of the generator is set at some middle frequency, such as 1,000 cps, and is suitably terminated in a load such as a resistor. Then both the voltmeter and the scope are connected across the load to read the voltage.

Now let's suppose that we want our scope "meter" to deflect, say, 10 divisions in either direction when the applied signal is 5 volts peak. But the reference voltmeter we are using may read only effective values. In this case the corresponding meter reading we want would be the effective value of 5 volts peak, or 5 $\times$ 0.707 = 3.535 volts.

Therefore, we adjust the generator output until the meter reads slightly over 3.5 volts. Then we adjust the VERTICAL GAIN control on the scope until we get the 10-division deflection desired. The scope is now calibrated, and we know that under these conditions every division on the grid represents

Continued on page 41

Are Loudness Controls Necessary?

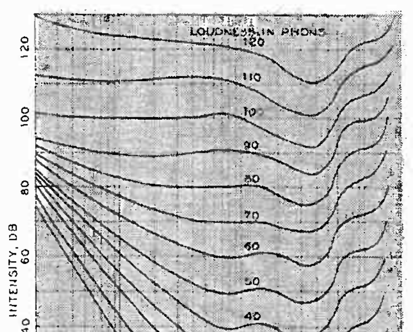
by George L. Augspurger

COMPENSATED volume controls have been used in commercial radios for years; it wasn't until someone linked the circuit with the names Fletcher and Munson that it became a subject of controversy among high-fidelity experts. The pros claim a loudness control is necessary to make reproduction scientifically natural at low playing levels, while the cons argue that such a circuit doesn't work in practice the way it should in theory. In this article we'll examine the facts which seem to favor some form of loudness compensation, survey the methods commonly used, and try a new functional approach to the problem.

Loudness Contours

Fig. 1 is the well-known *Fletcher-Munson graph of normal ear sensitivity.* These curves may be confusing because

Fig. 1. Average ear sensitivity curves.



they are drawn upside-down from customary representations of audio-equipment response curves. A frequency-response curve of a microphone, for example, shows upward peaks where the unit is most sensitive, and trails off toward the bottom of the paper at those frequencies which the mike can't "hear." In testing the ear's response, however, various test frequencies are adjusted in intensity until they sound as loud as the reference frequency, 1,000 cps. Then the actual frequency is measured and plotted on the chart to make up one graph line or curve. Thus, the point of *greatest* ear sensitivity is the *lowest* part of the curve, because less acoustic power was used to produce the rated loudness.

Neither your ears nor mine will register sensitivity curves identical to those used as a standard, but the general shape is close enough to represent anyone with normal hearing. The loudness we hear and the intensity of sound being radiated turn out to be two different things, so different units of measurement are used. Our ear's evaluation of loudness is measured in *phons.* The measured acoustic power is given in watts, and differences in acoustic intensity are rated in decibels. At 1,000 cps, intensity and loudness run hand-in-hand, and phons can be converted directly to decibels. At any other frequency this relationship doesn't exist.

The chart shows curves for 12 different loudness levels, beginning at 0 db (the threshold of hearing) and increasing in 10-phon steps. Note how the curve shape changes according to the loudness. To produce 20 phons at 100 cps, for example, takes far more acoustic power than to produce 20 phons at 1,000 cps. Yet for 100 phons it takes about the same acoustic power at each frequency.

The difference between acoustic intensity and loudness explains part of the effect known as *scale distortion.* Sound reproduced at a different level from that at which it was recorded will not only sound different in loudness; there will be an audible difference in timbre as well, because of the ear's different sensitivity curve at that level. Music played

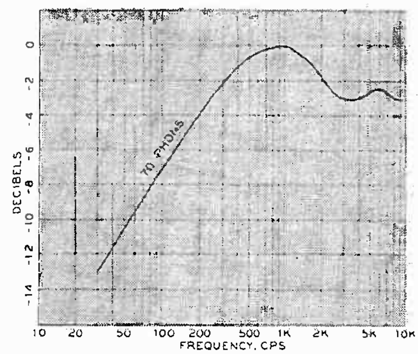


Fig. 2. Bass loss from ear response at 70 phons compared to 90-phon response.

at low levels becomes thin and lifeless. A speaking voice reproduced louder than normal is enhanced with "broadcast boom." Bass overemphasis in male voices is well known; but it is scale distortion of the opposite variety, that which makes soft music sound empty, which most concerns the audiophile.

This effect is illustrated in Fig. 2 which shows the effective *audible* response of a perfectly flat system, playing music 20 phons below the level at which it was recorded. Treble response has dropped 2 or 3 db, but bass zooms down rapidly to the threshold of audibility. The point is that even though the music is softer than it should be, we have the feeling that we should like to hear all the frequencies in the same loudness relationship as in the original. Although low-level playback without loudness compensation sounds much as though the orchestra were very far away, and is in that sense more natural, most of us would rather have the bass notes brought back to audibility. The treble loss is small enough to be barely perceptible, so that any compensatory boost is a matter of preference — it can't be justified by the Fletcher-Munson curves.

Loudness Controls

In an effort to compensate electrically for the increasing bass loss as volume is turned down, most amplifier manufacturers provide some sort of "loudness control" to take care of the situation.

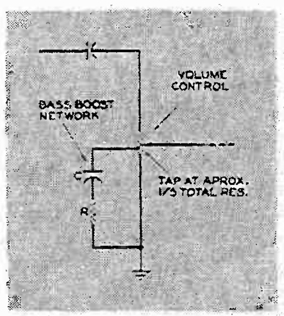


Fig. 3. Single-tap compensation network.

A loudness control reduces the bass frequencies in level more slowly than it reduces the level of middle-range frequencies, thus furnishing an automatically increasing bass boost as it is turned further down. The oldest form of such a circuit is the single-tap compensated control shown in Fig. 3. Such controls have been provided for years on the better radio sets. The single-tap control gives gradually increasing bass boost as volume is reduced until the tap is crossed by the moving contact. Below this point only intensity is affected. This simple control cannot approach accurate compensation — it is designed to give a pleasing effect, and that's all.

In 1948 the Livingston Corporation marketed a step-type attenuator called a Loudness Control. The Livingston circuit was similar to that of Fig. 4. This control has eleven positions, each one carefully based on the Fletcher-Munson curves. Assuming that everything else is set up properly, it gives scientifically accurate loudness contours no matter how quietly you play your phonograph.

A step-type control, though, is complicated and costly. It wasn't long before designs appeared for continuously variable controls. Some of these used multiple control taps; others tried ganged-potentiometer arrangements. The former is now sold as the Centralab Compentrol, and the latter is made by IRC. Both can be purchased at radio parts stores and can be installed in most amplifiers simply by replacing the existing volume control.

Because proper operation of a loudness control depends on the program material being reproduced, some means of altering loudness contours is usually included in commercial equipment. Harman-Kardon has a Contour Selector which is exactly what the name suggests. One position is provided in which the volume control has no frequency compensation at all. Several additional switch positions give a choice of varying degrees of automatic compensation, so the listener can choose the one that sounds most natural at the time.

A fifth type of automatic compensation has been suggested by Norman Crowhurst. "Why not," asks Mr. Crowhurst, "build a *dynamic* loudness-compensation circuit to boost bass automatically in inverse proportion to program level, regardless of whether loudness variations were caused by volume-control adjustments or by changes in the program material itself?" The effect would not be "natural," true enough; but ordinary loudness compensation isn't natural either, and listeners generally prefer it.

Loudness-Control Advantages

The five circuits just listed are the main types of automatic Fletcher-Munson compensation in audio equipment. Let's examine the advantages claimed for such arrangements:

1) Program material played at low level sounds much more alive and pleasing when suitably equalized according to Fletcher-Munson contours.

2) Trying to get the same results with ordinary tone controls requires resetting the bass each time volume is adjusted. Moreover, sufficient bass boost may not be available for low listening levels, or the bass-boost curves may not correspond in shape to those desired. The loudness control performs such tonal adjustments automatically, and the full range of tone-control action is available at any loudness setting.

3) Bass boost at low volume is accomplished within the control. No additional loss-type networks are employed; amplifier gain remains unaffected.

4) The user does not need to know anything about loudness compensation or Fletcher-Munson curves. Equalization is introduced completely automatically in the normal operation of the amplifier. No additional knobs or settings are needed.

Disadvantages

So much for the advantages of loudness controls. Now let's take a look at the planks of the opposing platform:

1) Whatever the characteristics of program material with loudness compensation added, it is decidedly *not* a natural effect. We are used to loss of bass as we move farther away from a program source, and when low frequencies are hauled back up to threshold level, we have the disturbing feeling that the sound is out of balance. In an effort to combat this, most loudness controls also provide 6 to 10 db of treble boost at low settings, although this is clearly not required by our hearing contours. The final sound is much the same as if we had used an ordinary volume control in the first place, except the middle frequencies are missing.

2) For loudness controls to work satisfactorily, additional knobs *do* have to be used. Automatic compensation is accurate only if the maximum position of the control corresponds to live-performance level. If it does not, the equalization provided will be audibly unbalanced. Consequently, additional "level set" or "contour selector" controls have to be adjusted each time the type of program material changes. If you're listening to a radio tuner, for instance, loudness compensation adjusted for music will be unlistenable for speaking voices.

3) If loudness controls are used incorrectly, especially by devotees of atomic bass, low-frequency boost from loudness and tone controls combined may overload the system. I have heard installations adjusted to produce pure cross modulation, much like the effect of talking through the blades of an electric fan. "I've got all the bass turned up, but it still doesn't seem to bring out the beat as it should."

4) If we are to hear bass at low levels, it must be reproduced at nearly

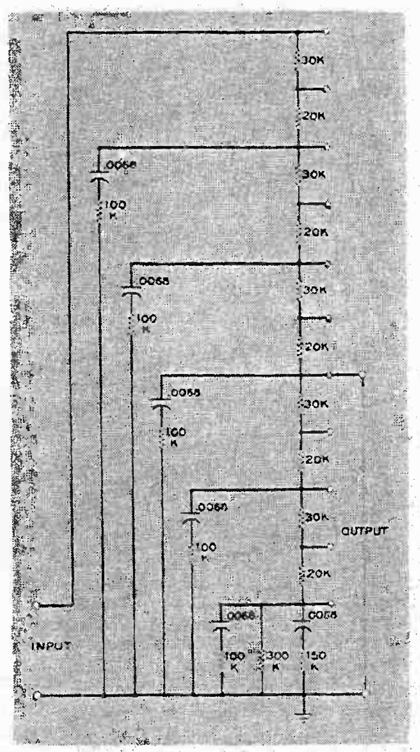


Fig. 4. A compensated step attenuator.

the same intensity as for loud listening. Therefore, with loudness compensation added, low-frequency speaker distortion becomes progressively noticeable as program level is reduced. 20% distortion at 40 cps (not uncommon for speakers

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BASIC ELECTRONICS

by Roy F. Allison

XVII: Diodes and Rectification

THE preceding chapter was concerned with electron emission from conducting materials, and the ways in which it can be accomplished. The heating method (thermionic emission) was found to be most practical for vacuum-tube cathodes. Physical construction of a vacuum-tube diode was described; this tube, with a heated electron-emitting cathode and a nonemitting plate, permits current flow through an intervening vacuum from cathode to plate but not in the reverse direction.

A diode's one-way conduction characteristic makes it useful in a great many electronic applications. Possibly its most common use occurs as a power-supply rectifier. Amplifying vacuum tubes require a source of relatively high DC voltage in order to perform their functions. This voltage may be furnished by a battery, as it was in the first days of radio and is now in portable radios. The battery (or group of batteries) used to provide high voltage was called the "B" battery; that used to power the filaments or cathode heaters was known as the "A" battery; and that used to furnish a small DC voltage for amplifying tube bias* was called the "C" battery. This nomenclature has been carried over into present-day usage, and any source of DC high voltage for vacuum tubes is referred to as a B supply. The voltage itself is called B-voltage or, because the output of the B supply is commonly positive with respect to ground, simply B+.

Batteries have a few real advantages as sources of operating power for vacuum-tube circuits, but they have drawbacks that are usually more important. They are heavy and bulky; they are not efficient in providing voltages higher than 150 volts, which many circuits require; and they must be replaced regularly, at some inconvenience and considerable cost to the user. These disadvantages become more severe as the power requirements are increased. With a modern hi-fi amplifier, for example, the expenses of battery replacement would be high enough to make any audio craftsman take up bridge. All high-fidelity equipments with vacuum tubes utilize rectifier power supplies, operating from the AC power lines.

AC line voltage, it will be recalled, reverses its direction 60 times each second, and has a sine-wave form. It is the main job of a power supply to convert this alternating voltage to a steady DC voltage of the proper value. Fig. 1 shows how we can make the first step toward accomplishing that objective.

First, we make one side of the AC line a common or ground point. The other side of the line is connected to a diode plate; the diode's cathode is connected through a load resistor R to the grounded side of the line. We have simply put a diode and a resistor in series across the line. The upper wave-form drawing (Fig. 1B) shows how the input voltage, that at point X, varies alternately positive and negative with respect to ground. During the positive half cycles of the input voltage, the diode's plate is positive with respect to its cathode, and the tube accordingly conducts current. The current flows through R from its bottom end to the top, making its top end positive. As long as the diode's plate is positive with respect to its cathode by any amount, the tube is almost but not quite a short circuit. Therefore, if R is reasonably large, almost the entire amplitude of the positive input half cycle appears across R and at the output terminals.

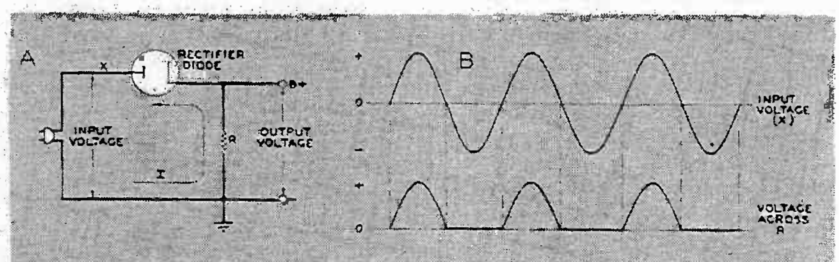


Fig. 1. A diode permits current flow in one direction only. If an AC wave form is applied, voltage across R is in a single directon: AC has been changed to DC.

When the voltage at X swings negative with respect to ground, the diode plate becomes negative with respect to its cathode, and conduction stops. No current flows through R, so the voltage across it — and the voltage at the output terminals — remains at zero during the negative half cycle of the input voltage. When the input voltage swings positive again, the diode conducts, and another positive voltage pulse appears at the output terminals. Since only half the input-voltage wave form is utilized, Fig. 1A is a *half-wave rectifier* circuit. The result is shown in the lower wave form, Fig. 1B: a pulsating but single-direction positive voltage.

This is basically and indisputably a DC voltage, because it is in one direction only. It is not a very desirable arrangement, though, as it stands. AC line voltages average about 115 volts RMS in amplitude, or about 163 volts peak value. Assuming a 13-volt drop in the diode because of its resistance while conducting, the peak value of the output voltage would be 150 volts. Now, the average (not RMS) value of a positive or negative half sine wave is 0.64 times the peak value; since each half cycle here is followed by a half-cycle interval of zero voltage, the average value is only 0.32 times peak value: $0.32 \times 150 = 48$ volts. This is too low a B voltage for most purposes. Further, the wave form is far from the steady DC voltage needed.

These objections can be overcome to some extent by putting a large capacitor in parallel with the load resistor, as Fig. 2 illustrates. The input voltage at point X swings above and below ground potential, as before. Capacitor C is uncharged and presents a very low impedance to a surge of current through it. The rectifier also has little resistance to current in this direction; as a result, an extremely large pulse of current flows on the first positive swing, charging the capacitor to the polarity shown. This charge voltage appears across the load resistance too, since it is in parallel with the capacitor. When the input voltage

*The subject of bias will be taken up in the following chapter.

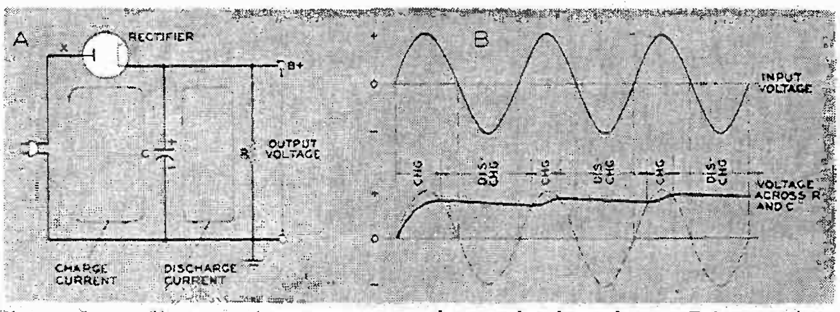


Fig. 2. Large filter capacitor evens out and smooths the pulsating DC wave form.

decreases below the value to which the capacitor has charged, conduction ceases because the cathode is now positive with respect to the plate. The capacitor then begins to discharge through the load, maintaining a slowly decreasing positive voltage across *R*. Because the capacitor is large (8 to 40 μfd are typical), it cannot discharge very much before the input voltage swings positive again. As soon as it exceeds the charge on *C*, conduction through the rectifier begins again, recharging the capacitor. Electrolytic capacitors are normally used; the plus and minus signs on the plates indicate their polarity.

The lower wave form in Fig. 2B, which is intended to show this filtering action clearly, is actually idealized slightly. Simply because the capacitor *is* so large, it takes quite a few cycles of the input wave form to charge it fully; the equilibrium portrayed in the second and third cycles would not be reached immediately. It should be realized also that *R* is intended to represent the entire load on the B supply — the effective or resultant resistance of all the circuits to which the supply furnishes power. If the load resistance is low (that is, if it draws a lot of current), the capacitor will discharge to a greater extent between positive peaks of the input voltage. That increases the variation in the B voltage (the *ripple*) and, at the same time, decreases the average B+ value. On the other hand, if the load resistance is very high, the ripple may be slight and the average B+ value may be nearly the peak value of the input voltage.

Ripple amplitude can be decreased for any given load resistance by increasing the size of *C*, the input capacitor. This cannot be carried too far, however, because the peak charging current may exceed the rectifier's rating. In any event, it is generally more effective to add other filter elements to the power supply, as shown in Fig. 3. Each successive series element in the filter (L, R_1, R_2) decreases the output voltage slightly, because the load current flows through it; but each reduces the remaining ripple by an amount greater than corresponding increases in the value of *C* (Fig. 2) could possibly do.

We have noted that changes in the resistance of the load cause changes in the B+ value. This is a *capacitor input* power-supply filter — that is, a capacitor is connected directly from the rectifier cathode to the common or ground terminal. It is this capacitor that is responsible for the power supply's sensitivity to load value, since the load current drawn determines how much the capacitor discharges between input-voltage cycles. Such a power supply is said to have poor *regulation*, and in some circuits it is important to have good regulation.

Regulation of a power supply can be improved in several ways. The simplest is certainly the bleeder method: putting an actual resistor of moderate value across the output of the supply, which draws a substantial proportion of the total current. By this means, the external load current may vary appreciably without changing the *total* current from the supply very much; it follows that the B voltage, although reduced in magnitude, remains more constant under changing external loads. The bleeder method can be used only when the total current of the bleeder resistor and the load does not exceed the rectifier's current rating, and when the waste of power in the bleeder can be tolerated.

Another regulatory method involves so-called *voltage-regulator* (VR) tubes across the output terminals. These gas-filled tubes maintain a practically fixed voltage across themselves, varying automatically the current they pass in order to do so (within a certain range of current values). This means of regulation is more precise than the bleeder method and requires less reduction of B voltage, but it is, of course, more expensive. Even more precise and more expensive regulation can be obtained by using ordinary vacuum tubes in special circuits.

A third way to improve output regulation is to remove the input capacitor, C_1, in Fig. 3. This converts the filter to a *choke-input* filter. All the charge current for C_2 must now flow through the choke; after equilibrium has been reached, the choke tends to keep a constant value of current flowing that is equal to the *average* value that would flow without any filter at all. Thus, although the output voltage is less dependent on the load, it is lower by far than with a capacitor-input filter.

With any of the circuits discussed so far, operating directly from the AC line, the maximum output voltage can only approach the peak value of the AC input. Still, such circuits are used in millions of table-model phonographs and radios, which do not require higher voltages nor very much power, and where cost is a powerful consideration. In these applications an additional advantage is obtained; the equipment is usable on a DC power line, provided the positive side of the line is connected to the upper input terminal (X). Although DC power lines are rarely found, this feature does make such equipment truly universal in application.

It is possible to obtain approximately twice the normal output voltage with a *voltage-doubler* circuit, such as that shown in Fig. 4, still using the AC line as a direct source of input voltage. During the negative half cycle of the input — that is, when the cathode of rectifier V_1 is negative with respect to its plate — capacitor C_1 is charged through V_1 to nearly the peak value of the input voltage. When the input voltage swings positive, V_1 stops conducting; the capacitor's right-hand plate is charged about 145 volts positive with respect to the left-hand plate. This charge is added to the positive-going line swing and to the total voltage across V_2 and C_2. As a result, C_2 charges to nearly twice the peak input voltage; say, to 290 volts. The output of the supply may be as much as 250 volts in normal circuits.

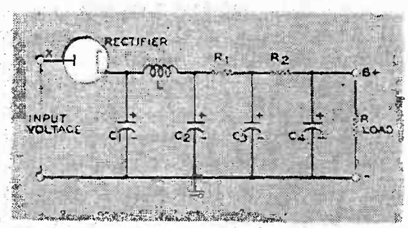


Fig. 3. Multisection power-supply filter reduces B+ ripple to negligible value.

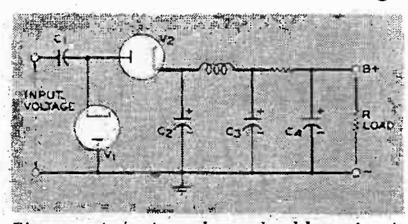


Fig. 4. A basic voltage-doubler circuit.

The voltage doubler has poor regulation, and requires a lot of filtering, but in some applications the regulation isn't too important. For medium-power equipments the voltage doubler is becoming more popular, particularly when it is necessary to keep costs down. The circuit is being employed extensively in new models of television sets which have generally used transformer-operated

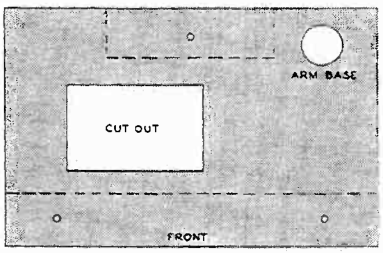
Continued on page 42

Turntable Level

The Audio Aid by Edward T. Dell, Jr., in the March issue of AUDIOCRAFT, provided for leveling a cabinet; but the following plan, used in my home-built equipment cabinet, makes it possible to level the turntable independently.

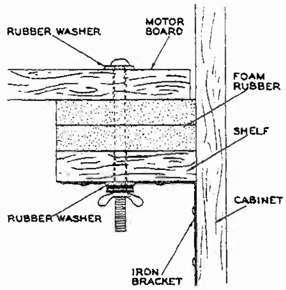
The inside dimensions of the turntable compartment were 14 in. by 21¾ in. The mounting board was cut from ¾-inch plywood, ⅛ in. less in all dimensions to allow free vertical movement. The depth to the top of the mounting board (in this case, 4¾ in.) was marked inside the compartment, and ¾-inch plywood shelves 3½ in. wide were installed on iron brackets inside the cabinet. One shelf extended across the front of the compartment, and a shorter one, 10 in. long, across the back. The depth of the tops of the shelves should be approximately 1¾ in. to 2¾ in. below the top of the mounting board, depending on the number of layers of foam rubber that are used to cushion the mounting board. Two 1-inch layers compress to about 1¼ in.

After making the cutout for the turntable bottom plate and drilling holes for mounting the arm, the board was placed



How the leveling screws are located on the turntable mounting board. Shelves beneath are indicated by dotted lines.

on the shelves, allowing 1/16 in. on each side, and three ¼-inch holes drilled through both the board and the shelf in a triangular pattern; two in front, 1¾ in. from the edge and 3 in. from each end, with the rear hole centered. The holes were reamed out so that ¼-inch bolts would have slight play. After mounting the arm, two 1-inch layers of foam rubber were placed on the shelves, the mounting board was placed on top, and 4-inch bolts were run through the three holes, piercing the foam rubber between board and shelf. Rubber washers were used at top and bottom, and iron washers and wing nuts below



Level as it appears in cross section.

the bottom rubber washer. These were tightened until the board was steady.

After the turntable was installed, a level was placed on it and appropriate wing nuts tightened or loosened until the turntable was level. This procedure can be repeated whenever necessary.

The arrangement is flexible, as the depth of the mounting board can be changed without much difficulty, and, if the turntable were replaced, a new mounting board could be installed easily.

Charles J. Bockler
Baltimore, Md.

Preventing Heat Damage

In the February 1957 Audio Aids section there is a helpful suggestion by a Mr. Zale about preventing heat damage while soldering. I can appreciate Mr. Zale's predicament in trying to use a pair of pliers for heat conduction.

In an effort to solve this same problem I checked with a local electronics supply house and was advised that General Cement makes a pair of long-nosed pliers that require pressure to open the jaws. The salesman told me he could order several and, in the same breath, made a suggestion that was awesome in its simplicity. He suggested using alligator clips.

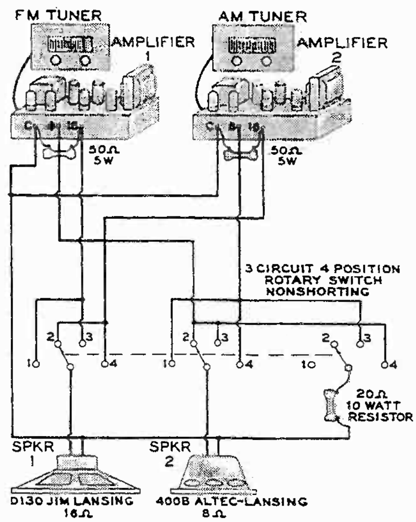
Since then I have used such clips in constructing several kits, one of which was an EICO VTVM containing a number of precision resistors. In the case of the VTVM, long-nosed clips are particularly useful because of restricted space where more than one resistor connects to the same terminal. Being smaller than the hemostat suggested by Mr. Zale, the clips do heat up some, but apparently they conduct enough heat to prevent damage to the components.

E. J. Henry, Jr.
Dallas, Tex.

Speaker Selecting System

Here's a simple speaker selecting system that can be used in a stereophonic radio or tape-playback setup to make it more versatile for monaural use. In this particular case, the main speaker is a 15-inch Jim Lansing extended-range unit in a bass-reflex cabinet. The secondary speaker is an 8-inch Altec Lansing in a smaller bass-reflex cabinet. Switching connections are shown in the drawing.

Position 1 of the switch connects the speakers to both amplifiers for stereophonic listening. In this position, either



Speakers connect to either one or both amplifiers by this switching system.

amplifier can also be used separately. Position 2 reverses the speaker and amplifier connections, and is used for connecting the main speaker (speaker 1) to amplifier 2 for straight AM listening. Position 3 connects both speakers to amplifier 1 for added depth when listening to FM only. Position 4 connects both speakers to amplifier 2 for the same effect with straight AM listening.

In the last two positions, the 20-ohm resistor is automatically connected across the secondary of the output transformer of the idle amplifier as a protective substitute load. The 50-ohm resistors, connected directly across the amplifier output terminals, are for protection of the output transformers while the switch is being operated.

L. E. Johnston
Madison, Wis.

Safety First!

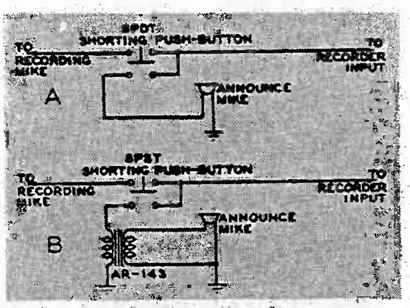
Some builders of amplifiers and old-time superheterodyne receivers have received nasty shocks from their handiwork; there have even been occasional fatalities. The hefty capacitors in power-rectifier circuits store enough energy to kill. They should all have bleeder resistors to discharge this lethal dose when they are disconnected from the power source. Wait, before prodding the innards of a hot chassis with a screw driver, for the bleeders to do their job. [*A suitable discharge resistor is one of 1- or 2-megohm value connected from the B+ output of the power supply to ground (chassis). Don't use a lower value unless you're sure that the extra current drain won't overload the power supply or upset the B+ voltages.* — ED.]

Another good rule is to keep one hand in your pocket and stand on an insulated surface when working with electronic equipment.

Harold G. Mitchell
New Philadelphia, Ohio

Tape-Recording Announce Mike

When making tape recordings, especially of a semiprofessional nature, it is often convenient to make a voice announcement before each number, for future identification. If the performance



Connections for tape-recording announce mike. Circuit A is for high-impedance, and circuit B for low-impedance mike.

is not under the direct control of the person running the equipment, it is usally not convenient to make these announcements with the recording mike.

An arrangement involving an extra mike located at the recorder, which could be switched on easily for these announcements, might be quite worth while. The setup described here has been used with good results.

Quality of reproduction is not important, so a cheap crystal microphone element can be used. If the recorder has a high-impedance mike input, use the arrangement in drawing A. The recording mike is cut out of the circuit while the announce mike is on — that is, when the push button is pressed.

A recorder with a medium- or low-impedance mike input can be used with the arrangement shown in drawing B. The transformer is a cheap miniature transistor transformer put out by Argonne and available from Lafayette Radio.

If a crystal microphone element (also available from Lafayette) is used, the device takes up very little space. It can often be mounted right inside the amplifier case.

This announce-mike arrangement permits the operator to announce each selection to prevent possible foul-ups later, when editing. Not every recorder needs one, but for those that do, it's very handy.

Paul Penfield, Jr.
Brookline, Mass.

FM Antenna Lead-in

In many locations, an outside FM antenna is necessary for satisfactory reception, even with the better FM tuners. Often, however, the expected increase in signal strength provided by the outside installation is accompanied by an unexpected and annoying increase in noise and interference from such sources as automobile ignitions and large electrical appliances. Usually, such interference is picked up by the lead-in, not the antenna. In many cases, the situation can be remedied by the use of a shielded 300-ohm twin line, such as Federal No. 111. Although the shielded twin line is more expensive than simple twin line, the interference rejection is often well worth the extra cost.

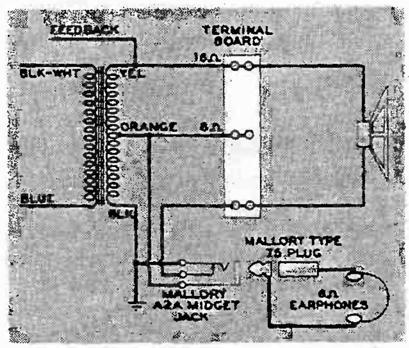
Louis R. Mills
Timonium, Md.

Headphone Jack

The never-ending battle of hi fi *vs.* TV when the two are in the same room has resulted, in my home, in the TV monster's being the victor. To salvage some use from my defeated hi fi, I have purchased a set of headphones to be used while the monster is bellowing forth.

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

To minimize the bother of disconnecting the 16-ohm speaker during headphone use, and utilizing the 8-ohm tap from my Dynakit amplifier for the 8-ohm headphones, a Mallory midget closed-type 2-circuit jack was put on the amplifier. The hot lead from the 16-ohm output transformer tap was left tied to one side of the speakers. The hot lead from the 8-ohm tap was tied



How headphone jack was connected to 8-ohm tap of the output transformer.

to the outside terminal board and to the outside terminal of the jack. The ground side of the speaker terminal board was brought to the single terminal on the jack. The shorting terminal was tied to ground. Connections are shown in the drawing.

With this arrangement, all that is necessary to listen through the headphones is to plug them into the amplifier. The speakers are automatically disconnected and the 8-ohm headphones are plugged into their proper impedance. This way it is not necessary to remember to switch from one to the other manually. The same type of jack was put on the TV for late-hour silent viewing.

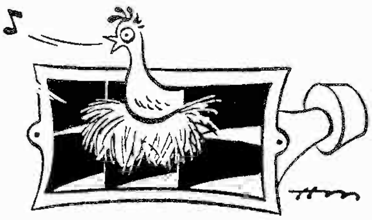
Albert J. Pezzo
Troy, N. Y.

Control Panels

Time was when the audiophile could get by with pencil lettering the knobs on his equipment to tell their function, or even dispense with marking altogether. Not so today. You never know who will turn the equipment on when you're not looking. Here's a way to make good-looking panels for items in which heat problems aren't involved.

A sheet of clear plexiglass and one of gold-colored plexiglass, both 1/16 in. thick (this material is available from most hobby and craft stores) are cut and drilled to the required dimensions. With a draftsman's lettering pen and guide, and a bottle of India ink, the words BASS, TREBLE, SELECTOR, etc., are inscribed on the gold plexiglass. To protect the ink from chipping, the gold panel is covered with the clear one. The result is a standard ⅛-inch panel of excellent quality.

Alex W. Thien
Milwaukee, Wis.



Sound-Fanciers' Guide

by R. D. DARRELL

Sneaking Up on Stereo

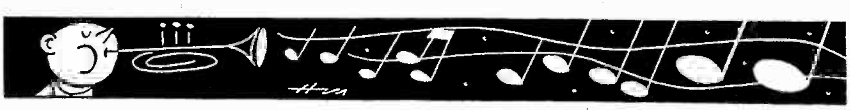
Observant readers of this column, and especially those who already own or hope soon to have home stereo systems, must have wondered why it has devoted such comparatively scant attention to stereo tapes. I have mentioned a few of them in passing, to be sure, and in terms which made no attempt to conceal the delighted fascination I find in them and the whole subject of this extraordinary new sonic medium, but I have hesitated to write about stereo at greater length here until I could be reasonably sure that such discussion and specialized reviews would be welcomed by more than a small minority of this journal's readers.

I still have no statistics on the followers of this department who boast stereo-sound facilities, but by this time it's certainly safe to assume that many of them would *like* to — if only they knew some practicable way of obtaining the necessary equipment and the nucleus of a stereo-tape library at reasonable cost. Well, "reasonable" is a term susceptible of widely varying interpretations, of course, but several current developments have begun to pull prices down to considerably lower levels than those which originally made stereo prohibitive for audiophiles of moderate means. A few relatively low-cost packaged systems are now available; but, more important, it has become possible to assemble components which will do a far better job for approximately the same money. While most stereo-tape releases are still priced stratospherically, a number of cheaper samplers have appeared recently which serve ideally not only as sonic illustrations of their manufacturer's catalogues, but also as convenient and versatile sources of program materials for checking the operation of new and experimental home stereo setups.

The subject of equipment itself is not properly the domain of this department and, in any case, I understand that it is to be explored in forthcoming articles elsewhere in these pages. Nevertheless, I should like to preface a survey of the present stereo sampler and demonstration repertory, by a few highly personal system generalities — which I trust you will liberally salt to your own tastes, since they are frankly idiosyncratic and in some points at least run directly against the main current of orthodox stereo thinking. In particular, I disagree strenuously with those propagandists and overenthusiastic converts who assert that *any* kind of stereo is better than even the best single-channel sound. Stereo at its finest is indeed superb and quite incomparable, but for me it never can realize its full potentialities (or even match first-rate single-channel reproduction) so long as it is restricted to two *small* speakers, which no hi-fi purist would ever consider adequate for ordinary LP playback purposes. These provide the inimitable stereo "effect" itself, all right, but in a curiously remote and bodiless manner, which has little if any of genuinely high-fidelity quality to my ears.

My ideal would naturally be stereo from two big, wide-range, two- or three-way speaker systems, but in actual practice I've found that when only one of these is available, the over-all response quality is not seriously compromised by the use of a smaller or less adequate speaker for the second channel. Whatever one uses, the two channels have to be reasonably balanced dynamically, of course, but in my view that doesn't mean that they have to be identical. As long as the channel with the big speaker system puts out a clean wide-range frequency spectrum, the restricted-range (but still clean) companion will supply the stereo effect satisfactorily. The over-all quality will be infinitely more acceptable (thanks largely to the nondirectional lows of the hi-fi channel) to discriminating ears than that produced by two identical narrow-range channels.

What this means to audiophiles who already own a good wide-range LP-playback system is obvious: you don't have to abandon or replace what you have, but can use its amplifier and speaker just as is for one (probably the left) stereo channel, while the other can be supplied by almost any reasonably good extra amplifier-and-speaker combination you happen to have on hand or can purchase inexpensively. In the former case, the additional requirements are a stereo tape deck and twin playback preamps with correct NARTB tape-playback equalization facilities — and even here, if you already have a tape player of some sort, it may be possible to convert it to stereo use without too great difficulty or cost. In this instance, it may be easier to install staggered heads, but in all others where you have a free choice before investing, I strongly recommend that you select stacked heads, if only for the reason that practically every stereo tape issued in a staggered-head edition is also available for stacked heads, whereas many others are now available in stacked-head editions only, and this disproportion is likely to become even greater in the future. (Unrepresentatively, most of the following samplers are issued in both versions; the few available for stacked heads only are indicated by an asterisk following the order number.)



First Stereo-Tape Choices

No matter what equipment you purchase, assemble, or build, when it is ready to play, the first thing you'll want to do is to check its operation in — and the best speaker locations for — your particular listening room with actual program materials. If you're a real engineering precisionist you may insist on dynamically balancing your two channels by means of a constant-frequency test tape; one of the previously available Ampex or Audio Devices 7½-ips full-track releases, or the just-announced special stereo alignment tapes by Sonotape (SWB AL 101, $11.95) and Stereophony Inc. (T 50, $1.98). I haven't received the latter two yet, but hope to soon and will of course promptly report on them. Meanwhile, however, it should be noted that neither they (nor the earlier full-track test tapes) are absolute necessities — except, of course, for checking the equalization provisions of home-built preamplifiers. Satisfactory enough channel balancing ordinarily can be achieved by playing any good stereo musical recording (or better, a series of these covering various types of music and performing ensembles), adjusting the individual channel levels by ear.

This implies, obviously, that individual channel-level controls are available — which is not always the case in some packaged stereo systems in the lower price ranges. If you have one of these, I'm sure you'll soon find it necessary to provide some means of remedying this lack (if only by relatively inaccessible chassis controls), for, by my experience, the necessity of depending on a single twin-channel volume control alone is frustratingly unsatisfactory. Not only are some individual level adjustments normally required to compensate for basic amplifier and speaker inequalities (even when these are purportedly identical for both channels) and specific room acoustics, but in many instances individual tapes — even among those produced by the same manufacturer — call for slight balancing readjustments to suit both the particular music at hand and one's personal aural tastes.

In such preliminary setup checks and tests, stereo sampler tapes are particularly useful, for not only are they usually considerably cheaper than regular full-length releases, but they normally range through a wide variety of musical and other materials. (In addition, of course, the samples themselves provide a helpful index to the specific tapings you are likely to want most — or to avoid — for your permanent library.) The following companies, in alphabetical order, have released such "appetizers" so far: Bel Canto (ST DX, 7-inch large hub, $3.95), Concert Hall (CHT/BN Dem 1, 7-inch, $4.00), EMC (STA 1*, 7-inch, $6.95), Livingston (LS 5-3 BN, 5-inch, $6.95), Omegatape (STD 6, 5-inch, now raised from its original $4.00 to $5.95, and STD 10, 7-inch, $5.95), Sonotape (SWB Dem 1, 7-inch, $6.95), Stereophony Inc. (C 80, 7-inch, $4.95), Stereotape (ST 1*, 7-inch, quarter-reel, $2.00), and Urania (UST D2, $3.98).

Of these, I have not yet heard the EMC entry — out for some time — or the just-announced second Omegatape (STD 10) and first Stereophony and Urania samplers, but I have tried out all the others and can commend them as extremely interesting and helpful despite (or in this case partly because of) their diverse quality and contents ranges. Some of them, especially the Bel Canto and Livingston samplers, are recorded at unusually high level and in some pieces with exaggerated stereo effect which makes them particularly effective on smaller stereo-playback systems, but which also gives a somewhat misleading impression of the dynamic levels and "fill-in" of the average full-length regular releases. The Bel Canto, Concert Hall, and Stereotape items (of those I have actually heard) are the only ones mercifully free from any vocal announcements or blurbs, but the narrations in the others are for the most part easily tolerable, while that in the Sonotape sampler is considerably better than tolerable. In all except the Bel Canto and Stereotape samplers (mostly popular or light music) and that from Concert Hall (which has only one jazz piece in an otherwise "serious" program), the jumps from pops to classics and back again are disconcerting — but so much the better for sonic variety's sake!

Of a somewhat different nature are the man-made and natural sound effects featured almost exclusively in the famous Concertape *Sound in the Round*, Vols. 1 and 2 (501 and 504, 5-inch, $7.95 each). These have been mentioned here before, but warrant praise again as ideal demonstrators of the stereo effect and its potentialities — particularly when one is trying to explain to a complete novice just what sonic dimensionality is all about. There are also briefer examples of similar materials (trains passing, etc.) included in the EMC and Sonotape samplers.

For completeness I should also mention the stereo demonstration tapes sponsored by RCA Victor (for stacked heads only) and the V-M Corporation (for staggered heads only), although these are not available for purchase, as far as I know, but are either given as bonuses with major equipment purchases or made available for dealers' demonstration use only. Short as the accessible sampler list may be at present, it is only a question of time before it is extensively augmented; and meanwhile, there is no more enticing repertory of first choices for what eventually may be (depending on your budget!) an impressive stereo-tape collection.

New Single-Channel Tape Samplers . . .

So much for stereo, for the moment. While I'm still in the domain of tape recordings I should go on to cover the latest samplers of the single-channel catalogues — that is, those released since my survey of last October. Four new entries have actually been heard and I've learned of the existence or imminent issue of several others. Of the former, Vol. 2 of Berkshire's *Highlights* (H 2, 5-inch, $1.50) is the most interesting for its all-serious musical contents (representing all twelve releases in Berkshire's second list), but most of these are exasperatingly brief, the recorded level sometimes quite low, and few if any of the pieces (mostly several-year-old Haydn Society and Urania recordings) are notable for outstanding technical excellence. Notably superior in this respect (as well as in its omission of vocal announcements) is the second Phonotapes-Sonore *Classical Sampler* (PM 3, 5-inch, $1.98), which presents longer excerpts or complete pieces from seven recent Vox recordings — Claire Coci's West Point organ album, Novaes' Chopin Waltzes, Van Remoortel's Grieg program, and others.

Omegatape's *Music for High Fidelity Shows* (D 8, 7-inch, large hub, $5.95) has preliminary and closing blurbs, in the rather hoarse and hollowly recorded voice of John Doucette. Some eighteen selections, mostly very brief, run a dizzying gamut from jazz and background music to such outstanding tape masterpieces as Scherchen's Bach *Brandenburg* Concertos, the Roger Wagner Chorale's Gregorian Chants, etc. The general technical quality, however, ranges from good to excellent. Miscellaneous as the materials may be, they do give a fine notion of the repertory scope of the latest Omegatape (Alphatape and Jazztape) catalogue additions.

Most striking of the present batch, though, and probably the most exciting release in this category to date, is Sonotape's first *Monaural Sampler* (Dem 2, 7-inch, half-reel, $3.50). This has not only the negative virtues of no announcements or blurbs and the positive ones of all more-or-less serious musical content, but also represents the highest technical standards yet established for 7½-ips single-channel tape recording and processing. Moreover, almost every one of the eight complete pieces or movements (Weinrich's Bach Toccata in D minor, Scherchen's Glière *Sailor's Dance*, Deutschmeister Band's *Radetsky* March, and virtuoso symphonic selections by Rodzinski) is a display work, dramatically calculated to demonstrate the full frequency and dynamic ranges, as well as the transient-response capabilities, of your tape playback system.

Other samplers not received for review include: Bel Canto *Dem Tape* (DT 27, $2.00) with ten excerpts and narration by Bill Steward; Celestial's *Artists' Review* (650, $6.75); Fantasy's sampler (75-1000, 4-inch, $2.95) featuring Brubeck, Mulligan, et. al.; Transition's *Jazz in Transition Sampler* (TRTP 30, 5-inch, $6.95), and the Web sampler (single track, $9.00). Two more, however — Omegatape's new *Jumbo Demo* (SPD 7, $5.95) and Jazztape's *Jazz for Reel* (D 9, $5.95) — are now on their way to me, I'm informed, and of course will be reviewed promptly.

Finally, a special *hors concours* com-

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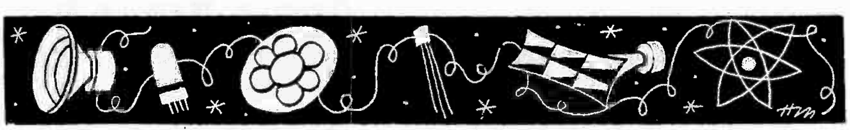


FM stations up-to-date

Changes and Corrections

THE following corrections and changes were received too late to be included in the FM station list published last month (pages 38-42, "FM Stations Up-to-Date"). We expect to have reprints of this list, including the changes given here, available soon. An announcement will be made in AUDIOCRAFT when the reprints are ready.

Corrections to Station List

CALIFORNIA, Ontario, KEDO: change ERP to .310.
San Bernardino, KVCR: change ERP to .770.
DISTRICT OF COLUMBIA, Washington, WWDC-FM: change LONG to 77.06.
FLORIDA, Jacksonville, WMBR-FM: change LAT to 30.31.
GEORGIA, Gainesville, WDUN-FM: change ERP to .340.
INDIANA, Evansville, WPSR: change LAT to 37.99.
IOWA, Davenport, WOC-FM: change LONG to 90.48.
MARYLAND, Baltimore, WJBC: change to WBJC.
MASSACHUSETTS, New Bedford, WNBH-FM: change ERP to 20.0 and delete DP.
OREGON, Eugene, KRVM: change ERP to .400.
PENNSYLVANIA, Altoona, WVAM-FM: change ERP to .360.
TENNESSEE, Chattanooga, WDOD-FM: change ERP to 42.0.
Knoxville, WUOT: change ERP to 70.5 and delete DP.
Memphis, WMCF: change ERP to 300.
WASHINGTON, Seattle, KISW: change LAT to 47.70.
— Note that **UTAH** stations were erroneously listed before and after **VIRGINIA** stations.

Construction Permits Added Between January 15 and April 3, 1957.

CITY	CALL	CH	(Mc) FREQ	(Kw) ERP	(Ft) HT	LAT	LONG
CALIFORNIA							
CP Los Angeles	KBBI	98	107.5	19.5	175	34.05	118.25
CP Palm Springs	new	21	92.1	1.0	–130	33.86	116.46
CP Pasadena	KPCS	07	89.3	.175	–563	34.14	118.12
CP Riverside	new	48	97.5	81.0	1690	33.97	117.29
MARYLAND							
C-CP Hagerstown	WARK-FM	95	106.9	2.2	225	39.63	77.71
CP Takoma Park	WGTS-FM	20	91.9	.010*	60*	38.99	77.00
MASSACHUSETTS							
CP Cambridge	WHRB-FM	96	107.1	.096	44	42.37	71.12
MISSOURI							
CP Kansas City	KCUR-FM	07	89.3	.450	115	39.03	94.58
NEW YORK							
CP Babylon	WGLI-FM	78	103.5	15.0	120	40.72	73.33
CP Troy	WRPI	18	91.5	.710	–120	42.73	73.68
PENNSYLVANIA							
CP Philadelphia	WXPN	05	88.9	.010*	173*	39.95	75.19
Pittsburgh	KDKA-FM	25	92.9	47.0	890	40.49	80.02

Call Letters Assigned

CALIFORNIA, Los Angeles, KDBX, CH 86.
MASSACHUSETTS, South Hadley, WMHC, CH 03.

Modification of Construction Permits (not yet on air)

CALIFORNIA, Los Angeles, KFSG-FM: change to KGLA.
Sacramento, KGMS-FM: change CH to 63, FREQ to 100.5, ERP to 9.8, HT to 160, LONG to 121.53.
Santa Barbara, KRCW: change ERP to 10.0, HT to –750, LAT to 34.42, LONG to 119.70.
Santa Clara, KSCU: change ERP to 1.1, HT to –55.

ILLINOIS, Carbondale, WSRV: change ERP to 21.0, HT to 390.
Chicago, WCLM: change ERP to 18.0, HT to 500.
INDIANA, Elkhart, WCMR-FM: change HT to 220.
VIRGINIA, Richmond, new: change to CP WRFK, FREQ to 91.1, CH to 16.

LAT and LONG only:

OHIO, Kent, WKSU-FM: LAT 41.15, LONG 81.35.
OREGON, Eugene, KWAX: LAT 44.05, LONG 123.08.
UTAH, Ephraim, KEPH: LAT 39.37, LONG 111.57.

Changes Requested for Existing Stations

CALIFORNIA, Los Angeles, KFAC-FM: ERP to 59.0, HT to 2820.
San Francisco, KEAR: ERP to 125.0, HT to 1100, LAT to 37.85, LONG to 122.50.
DISTRICT OF COLUMBIA, Washington, WFAN: ERP to 9.4.
GEORGIA, Atlanta, WAGA-FM: ERP to 36.2, HT to 965.
INDIANA, Indianapolis, WIAN: ERP to .890, HT to 45.
MAINE, Lewiston, WCOU-FM: ERP to 9.6.
MISSOURI, St. Louis, KCFM: ERP to 70.0.
VIRGINIA, Charlottesville, WINA-FM: ERP to .620, HT to 225.
UTAH, Salt Lake City, KSL-FM: ERP to 6.1, HT to 3660, LAT to 40.66, LONG to 112.20.

CP's Fully Authorized (now on air)

CALIFORNIA, Los Angeles: KXLU
Los Angeles: KHOF
COLORADO, Colorado Springs: KSHS
Denver: KTGM
DELAWARE, Wilmington: WJBR
INDIANA, Indianapolis: WFMS
MICHIGAN, Royal Oak: WOAK (call letters changed from WOAC)
MINNESOTA, Minneapolis: KWFM

Changes Granted Existing Stations

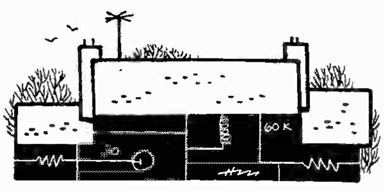
LOUISIANA, New Orleans, WDSU-FM
MASSACHUSETTS, Amherst, WAMF: CH to 08, FREQ to 89.5
Waltham, WCRB-FM: ERP also changed to 10.0.
NEW MEXICO, Mountain Park, KMFM.
TEXAS, San Antonio, KONO-FM.
VIRGINIA, Roanoke, WDBJ-FM.
WISCONSIN, Rice Lake, WJMC-FM.

Stations No Longer on the Air

CALIFORNIA, Pasadena, KWKW-FM.
San Mateo, KCSM.
INDIANA, South Bend, WHFS.
NEW YORK, Watertown, WWNY-FM.
PENNSYLVANIA, Chambersburg, WWPC.
TENNESSEE, Lenoir City, WLIL-FM
Paris, WTPR-FM.
VIRGINIA, Richmond, WLEE-FM.
WASHINGTON, Pasco, KALE-FM.
WISCONSIN, Madison, WIBA-FM.
ONTARIO, Kingston, CFRC-FM.
Kirkland Lake, CJKL-FM.
Sarnia, CHOK-FM.

Call Letters Changed

FLORIDA, Orlando, WKIS: changed to WORZ.
Palm Beach, WWPG-FM: changed to WQXT-FM.



SOUND FANCIER

Continued from preceding page

mendation should go to the newly launched series of aptly named *Cameos* that Phonotapes-Sonore has devised to meet the aching need for low-cost tapes of genuine musical value, which fall part way between sampler-length excerpts and full-length works or collections. Most of these are drawn from longer Phonotapes (for example, the four Norwegian Dances only, PMC 1004, from Van Remoortel's Grieg program). But the series will include also occasional selections appearing here for the first time on tape, like the pipe-and-drum tunes by the Black Watch Highland Regiment Massed Bands (PMC 1009) and the Six and Seven-Eighths String Band *Music of New Orleans* (PMC 1008, stemming from Folkways recordings). Each of these Cameos is an approximately half-length five-inch reel priced at only $2.98 — which should make them just about irresistible to every budget-conscious tape-library builder.

. . . And More Test Discs

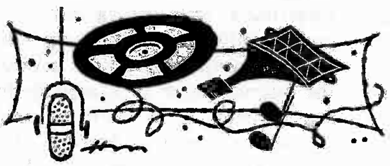
Since the foregoing tapes have left me such scant space for normal musical or demonstration LP reviews, I might use it more appropriately to deal with the latest additions (that is, since my extensive survey of last September) of true test recordings in disc form. For there are three current and highly noteworthy releases, representing both old and new contributors to this specialized repertory, on hand.

Emory Cook, returning to his once-favorite field of activity, offers the most novel work: a *Chromatic Scale Test Record* (Series 60, 12-inch LP, $4.98 with booklet), which presents for the first time materials that are not arranged in the usual mathematical sequences, but as frequencies corresponding to the standard musical tones ranging over an eight-octave chromatic spectrum. The same basic signals appear on both sides of this disc (tuning "A," chromatic progressions, rapid octave skips, bass and treble tone bursts). But on Side 1 these are recorded with Fletcher-Munson compensation for easier aural judgment of equal *loudness* response levels; on Side 2 the standard RIAA characteristics are employed, which, with proper equalization and flat response in the reproducing system, should produce uniform *intensity* output levels as measured by a VTVM or oscilloscope rather than by ear.

The notion here is a brilliantly imaginative one and the materials themselves are admirably devised to check both a wide variety of component-response characteristics and — perhaps even more significantly — those of your individual room's acoustics. It may be even more difficult than usual to interpret meaningfully the results obtained, especially by ear checks alone of Side 1, and there are surely test potentialities here which will demand long-term exploration before they can be fully utilized. For that very reason this presents a fascinating challenge to every audiophile determined to get the very best out of his home sound system and the extraordinary variety of pertinently *musical* testing materials provided here.

The new contributor is the Components Corporation of Denville, New Jersey, whose president, Jerry Minter, also doubles in brass by acting as commentator on one side of each of six different, specialized test discs (7-inch LP's, 89¢ each). The subjects covered are: *Wow and Flutter Too!* (No. 1106) with sustained 100- and 3,000-cps tones; *How's Your Stylus?* (1107) with reference and wear-test tones; *Quiet Please!* (1108) with standard and —60-db-level 1,000-cps tones and quiet grooves; *Tracking Special* (1109) with a continuous 100-to-10-cps glide tone; *Vertical/Lateral Response* (1110); and *What! No Hum?* (1111). These vary considerably in both quality and usefulness: the first four are reasonably conventional; the fifth strikes me as a not particularly meaningful — and unduly severe — test of pickups; but the last (with its sonic examples of various types of hum and its series of 58-cps tones, for beating against system-generated 60-cps hum, at various levels down to —60 db) is a real gem. The commentaries themselves are extremely illuminating and helpful, but I could have easily done without the occasional plugs for the manufacturer's own turntables and "hydrafeed" recording lathes (fine as those products are), and even the low price of these discs scarcely excuses the omission of plastic inner envelopes. Several of my review copies were quite gritty (intolerable in a test record above all), and since the pressings themselves looked like the characteristically fine products of the Microfusion process, I can ascribe this only to dirt resulting from the use of a loose manila envelope alone.

Components Corporation also has prepared the *Hi-Fi Test Record*, a twelve-inch LP which is obtainable solely by mail — but at the tempting price of



only $2.50 — from *Popular Science Monthly*, Dept. 357, 353 Fourth Ave., New York 10, N. Y. Its conventional but comprehensive materials (vocally announced where necessary as well as described in an accompanying instruction leaflet) include outer- and inner-groove maximum-level 1,000-cps signals for pickup checking, quiet grooves, average and —60-db-level 1,000-cps tones, a 400-cps "A," maximum- and normal-level glide tones (20 Kc to 20 cps and 15 Kc to 30 cps respectively), repeated 300-to-20-cps glide tones for tuning speaker enclosures, and a series of twelve spot frequencies for RIAA equalization checking. In addition there is a "Sound-Effects Quiz" (with answers in the accompanying leaflet) and five 1890 Regina Music Box selections, apparently thrown in for variety's sake mainly, although they do incidentally provide some extremely crisply recorded transient-response check materials.

FILL 'ER UP

Continued from page 22

replace seats and the other panels and trunk lining removed when you began this project. At least the stuff will be out of the way while you're attaching the extension up front to the car radio.

Installing the control switch is fairly easy. Most radio wafer switches are not designed to handle heavy speaker currents, so it's wise to buy a speaker control switch identified as such. These will have clear installation instructions.

Phasing the extension speakers to the car radio is done by holding bared ends of the extension wire against wires to the dash-panel speaker. The connection that does not reduce the volume of the cowl speaker is the correct connection, and the coded wire must be traced through the three-way switch or fading control to the proper lug of the cowl speaker. Remember, if the volume falls off when the extension is plugged in, the connection is backward. Reverse the wires and everything will be fine.

Speaker installations in the head lining, kick panels, or under the seat can be made as easily. If there is no solid metal you can bolt the speaker to, use a piece of ½ or ¾-inch Firtex, Celotex, or other soft fiber board in the same manner as the Masonite hardboard used for the rear-speaker installation. Then pad behind the speaker with Fiberglas to absorb rear-cone radiation. Grilles can be mounted with the speaker and the whole panel-and-speaker assembly pushed into place, or the grille can be screwed to the panel and fiber-board backing after the panel has been reinstalled. Use your imagination, and have fun; for the basis of success in this project lies in good dispersion of sound sources throughout the car. The result is a sound system unlike that of any automobile radio you've ever heard.

LOUDNESS CONTROLS

Continued from page 31

run at high levels) may be masked at live-performance intensity by the middle frequencies, but is quite distressing when the middle frequencies have been dropped 15 to 20 db.

A New Approach

Comparing the pro and con arguments thus far, we might reach the conclusion that, while some sort of automatic compensation is desirable, the goal of accurately and automatically obtaining optimal compensation can be achieved only if our hi-fi set is connected to an IBM calculator complete with built-in music critic and recording engineer. One mistake seems to have been made in succumbing to the convenience of continuously variable accurate compensation. An honest listening test demonstrates that the old single-tap control *sounds* almost as effective.

Suppose we approach the problem from the functional angle, examining the ways in which we ordinarily use our hi-fi installation, under what con-

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LOUDNESS CONTROLS

Continued from preceding page

ditions loudness compensation is desirable, and what type of compensation is best under these conditions. Most of us will find that we listen in two distinct ways:

A) *Serious listening*, with the volume set anywhere from, "Can't you turn it down a little, dear?" to live performance levels.

B) *Background music*, for reading or parties or other occupations. The volume is set just loud enough to be soothing but not distracting.

Doesn't almost all of your listening fit into one of these headings? Serious

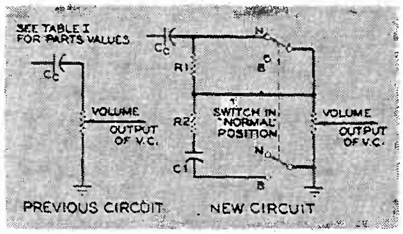


Fig. 5. Simple circuit boosts bass in background-music position of the switch.

listening simply isn't done at low volume. Can you imagine anyone listening intently to Beethoven's Ninth at a barely perceptible level? Yet this is a situation for which commercial loudness controls are designed.

Since most of us will agree that our hi-fi set is used either for serious listening or background music, the logical type of loudness compensation is a circuit which can be adjusted to these two con-

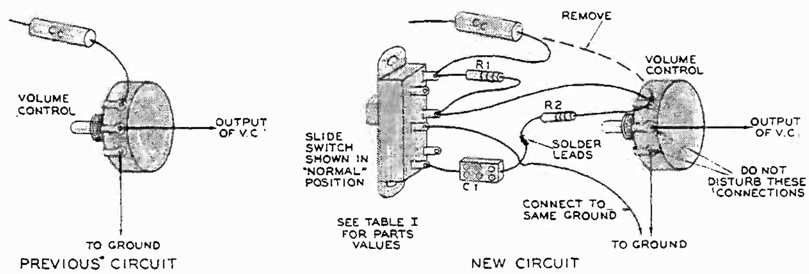


Fig. 6. How to interconnect component parts of the author's compensation circuit.

ditions. For serious listening, a loudness control is a nuisance. At any normal level we adjust bass and treble controls anyway, and there is more than enough bass variation available to give us realistic reproduction. For background music, however, we may drop the level 20 db or more. We aren't interested in natural sound now, and considerable bass boost is needed to fill in the thin effect. No treble boost is required because a "soothing" sound is what we want.

Suppose we include in our amplifier a single two-position switch labeled NORMAL-BACKGROUND. In the NORMAL position the volume control has no loudness compensation; we adjust bass, treble, and volume as usual. For soft music the switch is flipped to BACKGROUND. In this position suitable bass boost is added below 1,000 cps and the level is dropped 20 db at the same time. Bass, treble, and volume are left in their normal positions. The function switch thus sets the system for either of two modes of operation, and the other controls are merely touched up in either position.

The Circuit

Fig. 5 is the schematic diagram of such a dual-function loudness circuit, and Fig. 6 is the corresponding pictorial diagram. Two resistors, a capacitor, and a double-pole double-throw switch are the only components needed in addition to the existing volume control. Table I gives

Table I

Vol. Cont. Res.	R1	R2	C1
100 K	100 K	10 K	.02 μfd
500 K	470 K	47 K	.004 μfd
1 Meg	1 Meg	100 K	.002 μfd

parts values for common volume-control resistances. The amount of volume change and bass boost was established empirically after trying different component values for about two weeks.

If the amplifier volume control is in the output circuit of a cathode-follower stage, circuit dress isn't too important in making the modification. In high-impedance circuits, on the other hand, the new switch should be mounted as close to the volume control as possible to guard against hum pickup. Switching is so designed that either a shorting or nonshorting switch may be employed without clicks or pops.

I have two hi-fi systems in my home. The dual-function switch described in this article is part of the big experimental set. A commercial loudness control was originally included in a smaller unit built for portability. It was not until I had ripped into the smaller set for the third time in an effort to make the loudness control listenable that I started experimenting with a function switch instead. It has proved to be a logical and simple solution to the desire for workable loudness compensation.

FREE ENERGY

Continued from page 23

many circuits of which have been published. One arrangement is shown in Fig 2. The work horse of transistor experimenters, a CK722 junction transistor, is connected in a grounded-emitter oscillator circuit. With the transformer impedances as shown, and the antenna length reduced to 10 ft., current readings were as follows: emitter current, 16 μa; collector current, 16 μa; and base current, 2 μa.

Although published oscillator circuits of this type have high- and low-impedance transformer connections, as

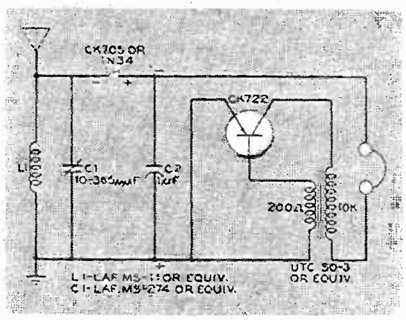


Fig. 2. A transistor audio oscillator operated by "free-energy" power supply.

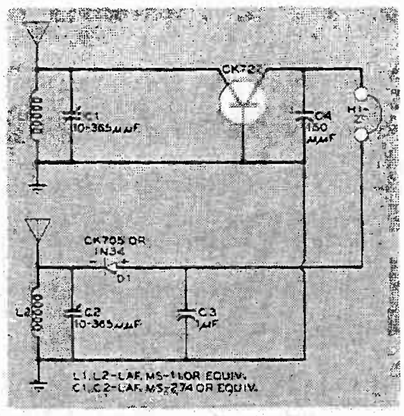
shown in the diagram, I have found that a purer tone of greater magnitude can be obtained by lowering the impedance in the collector circuit. In fact, a good audio signal was obtained with the collector circuit winding reduced to 500 ohms. This occurred not only with the free-power rectifier but with power obtained from a flashlight battery. When the impedance of the collector winding was increased beyond 20,000 ohms, there occurred a periodic blocking of the oscillator, the rate of which decreased with increasing transformer impedance. At 50,000 ohms impedance, oscillation ceased.

Several transformers were used that happened to be on hand. The experimenter may use UTC SO-3 or SSO-3 transformers which have been employed extensively for transistor experiments.

Frequency of the audio oscillation varied between about 500 and 900 cps with tuning of the resonant circuit by capacitor C1. Adjustment of C1 changed the rectified voltage available to power the transistor oscillator circuit, because it detuned the RF circuit from the carrier frequency of the radio station. The oscillator output was, of course, greatest when the circuit was tuned exactly to the AM station frequency, and the pitch of the tone was then lowest. Tuning off the station frequency resulted in less output and higher audio tone. Oscillation continued until the emitter current dropped to 8 μa.

The circuit of Fig. 1C was also used

to power a simple transistor receiver, as shown in Fig. 3. Another loopstick and tuning capacitor were connected to a CK722 amplifying and detecting stage; this tuned circuit was driven by a separate short antenna wire. Sensitivity was good and signals were received quite well. The selectivity of this tuned circuit, however, was rather poor, and a more selective front end may be required in certain localities. Many different types of tuned circuits can be found in the literature.



Figs. 3A, above, and 3B, right. Schematic and pictorial wiring diagrams for a transistor AM radio tuner. Operating voltage is furnished by RF rectifier.

A push-pull power-supply circuit was tried which provided greatly improved selectivity. This is shown in Fig. 4. Sensitivity was not quite so good as with the loopstick circuit, but there was plenty of power to operate simple transistor circuits. Under the same test conditions as for the circuit of Fig. 1, current flow through the load resistor was 165 μa.

The coil used in this circuit was wound on a 1-inch-diameter form. If a plastic or bakelite tube is not handy,

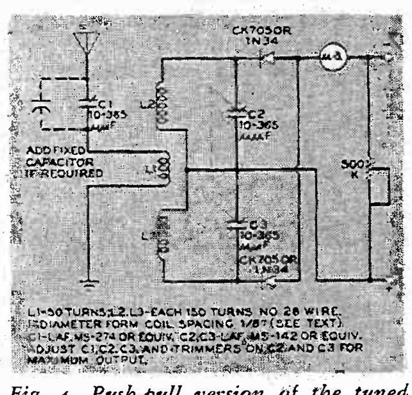
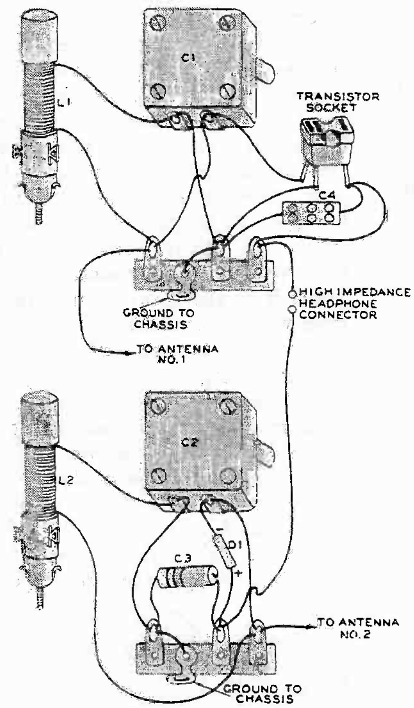


Fig. 4. Push-pull version of the tuned radio-frequency rectifier power supply.

a cardboard tube may be used with a couple of layers of waxed paper wrapped around it and held in place with cellulose tape. L1 is spaced 1/8 in. from both L2 and L3. All sections must be wound in the same direction. After winding the coil, I realized that it would be a useful item to add to my collection of transistor circuit parts, so I cut some scrap plastic sheet and, with a little heating, formed it into mounting brackets which were cemented to the coil form as shown in the photograph.

A push-pull coil of higher efficiency might be developed with two of the loopsticks. They could not be placed back to back, though, because the wind-



ings would be in opposite directions. They could be modified by cutting off the windings of one loopstick and turning them around, securing them in place with cement. The two loopsticks could then be cemented together with a piece of 1/4-inch wooden dowel stick, which fits snugly into the loopstick coil form. The dowel should be long enough to hold 40 or 50 turns of wire (which would comprise L1) between each loopstick. I haven't actually tried this, but there seems to be no reason why it wouldn't work.

It should be realized that, in the case of a simple transistor receiver, the station listened to may supply the power to operate the receiver, or power may be recovered from one station to operate the receiver tuned to another station. Some other interesting applications of rectified RF power for transistor circuits will probably occur to other experimenters, and we will, no doubt, be seeing more magazine articles in the future on various means of using this free power.

TEST INSTRUMENTS

Continued from page 29

0.5-volt peak. This is true for any wave shape, and at any frequency within the specifications of the scope.

A simpler method of calibration, though somewhat less accurate, is possible on those scopes having a built-in calibrating voltage. In this case, the entire calibration process is self-contained, and the generator and meter are not needed.

It was mentioned last month that some scopes have a 1-volt peak-to-peak signal appearing at an external binding post. This may in turn be connected to the VERTICAL INPUT binding post, and the VERTICAL GAIN control adjusted for any desired deflection. Remembering that 1-volt peak-to-peak is only 0.5-volt peak, we can see that, for our previous example (5-volts peak in 10 divisions), we would adjust gain for a deflection of just one division in either direction.

But now suppose we want the scope to indicate 1-volt RMS at 15 divisions. Then 0.5-volt *effective* would deflect to 7½ divisions. But since our standard signal is 0.5-volt *peak*, it should deflect the beam $7\frac{1}{2} \times 1.414 = 10.6$ divisions. Then with our VERTICAL GAIN control set for a deflection by the test signal to a little over 10½ divisions, we know that a signal of 1-volt effective value will deflect the beam 15 divisions in either direction from the zero position.

This type of calibration is limited by the accuracy of the power-line supply voltage, as well as the normal tolerances in the component parts. Within these limitations, however, it is quick and self-determining, without any external equipment required. Whichever calibration system is used, it should be followed each time the equipment is to be used in this manner. Even if no adjustments have been changed, these variable factors mentioned, plus others such as tube aging, may tend to impair the accuracy of the readings after the passage of time.

But the use of the oscilloscope as a voltmeter is only the beginning of the story. When we use this instrument to examine wave forms, we can learn nearly everything we need to know about the performance of audio equipment.

As we already know, it is usual practice to apply the signal under test to the VERTICAL INPUT terminals of the scope. By the use of amplifiers, this voltage causes a vertical displacement of the electron beam striking the face of the tube. At the same time, we have a linear time base in the form of a sawtooth voltage within the instrument, which sweeps the beam horizontally across the tube at a given rate. The frequency of this sweep voltage is often adjusted to ½, ⅓, or ¼ that of the signal under test. Under these conditions we will see 2, 3, or 4 complete cycles of the test signal.

To test the performance of an audio amplifier we use the setup shown in Fig. 3. The audio-frequency generator (see AUDIOCRAFT, June and July 1956)

Continued on next page

TEST INSTRUMENTS

Continued from preceding page

should be capable of producing a pure sine wave with negligible distortion. Signal level should be comfortably within the rated input capabilities of the amplifier. The load resistor should be equal to the output impedance of the amplifier.

Now let us examine a few typical scope patterns and see what they tell us about the amplifier's performance. In Fig. 4A we see serious flattening or "clipping" of one peak, in this case amounting to about 10% harmonic distortion. One cause of this could be excessive grid bias, resulting in one or more tubes being driven beyond cutoff on the negative half cycles. Alternatively, either the grid bias or the plate voltage might be too low, or an inoperative tube in a push-pull stage could cause clipping of either negative or positive peaks, depending on its position in the circuit.

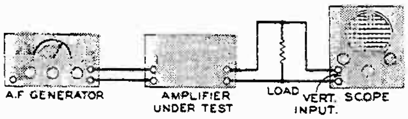


Fig. 3. Using scope for amplifier test.

Fig. 4B shows equal distortion of both peaks. In this case both second and third spurious harmonics are present. Fig. 4C shows clipping of both peaks, usually an indication of overload.

The scope pattern in Fig. 4C begins to resemble the rectangular wave form commonly called the square wave. This pattern, while extremely undesirable when due to amplifier distortion, can be most useful in analyzing amplifier performance and distortion. In other words, when we feed a sine wave into an amplifier, that is the shape we want to see at the output. But when we deliberately feed a square wave to the input, the resulting scope pattern across the output can give us information which otherwise would require much lengthier testing.

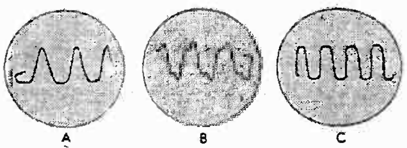


Fig. 4. Appearance of several types of distortion as seen on the oscilloscope.

The square wave, shown in Fig. 5A, is so useful to us because it is so very complex. We have seen from Fig. 4 how the addition of harmonics to a simple sine wave can drastically alter its wave shape. Adding more and more odd-harmonic sine waves to a sine-wave fundamental brings the resulting wave form nearer and nearer the shape of a square wave. Finally, the resulting wave becomes a perfect square wave if an infinite number of odd harmonics is added to the fundamental.

So if we choose a fundamental square-wave frequency near the bottom of the audio range, we have here in one convenient wave form, and all at once, a signal which covers the entire audible spectrum! Thus we can learn in a single observation the performance of the amplifier not only at the test frequency, but also at many points considerably removed from the fundamental.

In the theoretically perfect amplifier, the output wave form is an exact duplicate of the input, except at a higher power level as determined by the amplifier gain. In the case of the square-wave input, this requires some doing. If the high-frequency performance of the amplifier is excellent, the front of the wave form will be vertical and sharp-cornered, as shown in Fig. 5A. But if the high-frequency response is not so good, the rise of the pulse may tend to lean forward, and the front corners will become round-shouldered, as shown in Fig. 5B. This type of trouble may also be accompanied by amplitude distortion or phase shift, or both.

The flat top portion of the wave, on the other hand, is the clue to the bass response. As long as it is truly flat and horizontal, as in Fig. 5A, all is well. But if the tops of the pulses begin to look droopy or to sag from the horizontal (Fig. 5C), then the bass response slopes below the fundamental frequency of the square wave. Even slight bass rolloff produces a marked slope in the wave.

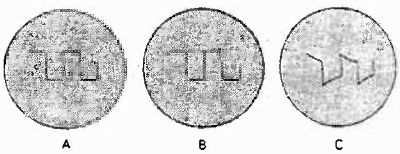


Fig. 5. Square-wave deformation visible on scope shows response characteristic.

We have now seen how the versatile cathode-ray oscilloscope can be used to measure AC voltages, both steady and transient, and how it can show up and analyze several common types of distortion, using both sine and square waves. Next month we'll continue our discussion of amplifier performance and distortion analysis while using the scope.

JENSEN-CABINART

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because they insist (rightly) that differing test conditions make speaker specs virtually meaningless, we should say that the Concerto 12 two-way system outperforms many three-way systems. The natural resonance frequency of the woofer, we found, was 48 cps. Installed in the enclosure, and with the enclosure in a corner of our standard test room, bass was full, even, and pure down to about 42 cps, where a trace of harmonic appeared. By 30 cps the fundamental tone had almost disappeared, but so had the harmonic response, which is desirable. Judged by ear, response was smooth upward throughout the middle range, with an evident increase in output from 3 to 4.5 Kc; full output was maintained up to 11 or 12 Kc, and there was still audible response above 15 Kc. Dispersion from the compression-driver tweeter was unexpectedly good to above 10 Kc, and fair at 13.5 Kc.

Efficiency of the tweeter is far greater than that of the woofer, which is why the high-frequency balance control is furnished. Excellent instructions for properly adjusting this control are given. The best adjustment will vary with the room, but not by a great deal; for us it occurred with the control-knob arrow pointing between "L" and the second "A" in the word "Balance" inscribed on the balance-control escutcheon.

The over-all system efficiency is close to average. Our general impression is one of tight and full but not oppressive bass, well balanced by a slight brightness; a most satisfactory sound, that can be listened to for a long time without weariness. We preferred the balance obtained with the enclosure in a corner.

Since we have grown to detest the sight of sandpaper and steel wool, the P-207 prefinished dress kit was a joy to use. You simply staple a single piece of grille cloth all around the enclosure (except for the back panel), from bottom to top. The base and base molding arrive in the carton as an assembled and beautifully finished unit that slides over the bottom of the enclosure, and is secured to it by four screws driven up into the bottom panel. The molding conceals the bottom edge of the grille cloth. Turning the cabinet over, you slide on the finished top and upper-molding assembly in the same way. It is held by four screws driven up into it through the enclosure's top panel. That's all there is to the job!

If you've been impressed (as many have) with a factory-assembled Jensen speaker system, these Jensen and Cabinart kits provide an inexpensive means for you to own its equal in appearance and performance, with a small investment in labor that is more fun than work.

BASIC ELECTRONICS

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power supplies in previous years. A voltage doubler cannot be used on a DC power line, unfortunately.

All these relatively low-voltage power-supply circuits can be equipped with metallic-oxide or semiconductor rectifiers; germanium diodes are now available for use in TV power supplies, and show great promise in this application. These devices are generally more reliable and longer-lasting than vacuum-tube rectifiers, and until recently were considerably more expensive.

Even a voltage-doubler circuit cannot furnish the high B+ voltages often needed in high-fidelity equipment, however. Here also it is important to reduce ripple to the absolute minimum, to keep hum voltages out of the system, and this is difficult to do with the circuits described so far. Finally, it is never good practice to use one side of the power line as a common terminal if it is possible to avoid doing so. These problems can all be overcome simultaneously by using a transformer-operated full-wave rectifier power supply, which can be seen in typical form in Fig. 5.

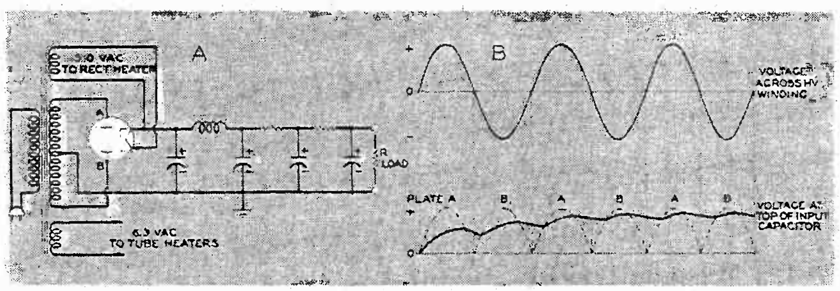

Fig. 5. Typical full-wave rectifier power-supply circuit, operated by transformer.

The transformer has three separate secondary windings: a 5-volt RMS winding for the rectifier-tube filament or heater, which most rectifiers of high power rating require; a 6.3-volt winding to furnish heater power to other tubes used in the equipment; and the high-voltage secondary winding. This winding is tapped at the center, and the voltage across the entire winding may be any value from 125 to 1,200 v RMS, depending on the application. A typical value is 750 v. This means that the voltage from *each* end of the winding to the centertap is 375 v RMS, or about 530 v peak.

As Fig. 5 shows, this centertap is connected to the common terminal, or ground. There is no connection whatever to either side of the AC power line. Opposite ends of the high-voltage winding are connected to individual plates of a full-wave (two-plate) rectifier; a common cathode or filament (depending on the rectifier type) serves both plates. When the top end of the HV winding is positive with respect to the centertap, rectifier plate A is positive with respect to the cathode. A surge of current flows out of the grounded centertap into the bottom plate of the input capacitor, out of the top plate of the capacitor to the rectifier cathode, from the cathode to plate A, and through the top half of the HV winding to complete the circuit. After the charging peak passes, the input capacitor and the rest of the filter circuit serve to keep current flowing steadily in the load.

Now, while the top end of the secondary winding was positive with respect to the centertap, the bottom end was negative. In consequence, rectifier plate A could conduct while plate B could not. When the input voltage swings the other way, plate A cuts off, but plate B becomes positive with respect to the cathode and begins to conduct. The route this time is from the centertap to the lower plate of the input capacitor, out the upper plate to the rectifier cathode, from the cathode to plate B, and through the lower half of the HV winding back to the centertap. Note that the capacitor charge direction was the same in each case. In this full-wave circuit the input capacitor is recharged twice during each input-voltage cycle, as diagrammed in Fig. 5B, so that it does not have a chance to discharge as much as in a half-wave circuit. Accordingly the ripple voltage is smaller at the beginning of the filter, and its frequency is twice as high; it is much easier to filter out. Also, the output voltage can be of virtually any magnitude desired, because it can be controlled by the number of turns used in the transformer's high-voltage secondary winding.

The remarks on output-voltage regulation made previously for half-wave circuits apply also to full-wave power supplies.

TRANSISTORS

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gram of Fig. 2, along with the following relationships:

$$R_{ie} = \frac{V_{eb}}{I_b} \qquad \ldots(10)$$

$$R_{oc} = -\frac{V_{ec}(R_l = \infty)}{I_c(R_l = 0)} \qquad \ldots(11)$$

$$A_{ve} = \frac{V_{ec}}{V_{eb}} \qquad \ldots(12)$$

$$A_{ie} = \frac{I_c}{I_b} \qquad \ldots(13)$$

$$g_{se} = \frac{I_c}{V_g} \qquad \ldots(14)$$

$$G_e = -A_{ve}A_{ie} \qquad \ldots(15)$$

$$G_{te} = 4g_{se}^2R_gR_l \qquad \ldots(16)$$

where the grounded-emitter current gain is denoted by adding the subscript *e* to A_i, etc.

Solving Fig. 2 for these relationships is not easy, although it is done by algebra alone. No useful purpose would be served in putting the derivations down here; the reader can check the results below, and indeed would benefit by so doing.

Certain things, however, become obvious just by looking at Fig. 2. For example, consider the input resistance R_{ie}. This will be a function of the load resistance R_l. Thus, when $R_l = 0$, we know that the output voltage $V_{ec} = 0$, so the fictitious generator $h_{re}V_{ec}$ is equal to zero. In this case, the input resistance is just h_{ie} in value. And when R_l is "small enough" we expect the same value.

If the reader goes through the steps of solving the circuit of Fig. 2 for the ratios in equations 10 through 16, he should obtain

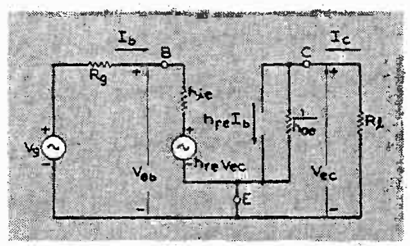

Fig. 2. Quantities in h-parameter terms.

$$R_{ie} = h_{ie} - \frac{h_{fe}h_{re}R_l}{1 + R_lh_{oe}} \qquad \ldots(17)$$

$$R_{oe} = \frac{h_{ie} + R_g}{R_gh_{oe} + h_{ie}h_{oe} - h_{fe}h_{re}} \qquad \ldots(18)$$

$$A_{ve} = \frac{-h_{fe}R_l}{h_{ie} + R_l(h_{ie}h_{oe} - h_{fe}h_{re})} \qquad \ldots(19)$$

$$A_{ie} = \frac{h_{fe}}{1 + R_lh_{oe}} \qquad \ldots(20)$$

$$g_{se} = \frac{h_{fe}}{(R_g + h_{ie})(1 + R_lh_{oe}) - R_lh_{fe}h_{re}} \qquad \ldots(21)$$

$$G = \frac{h_{fe}^2R_l}{(1 + R_lh_{oe})[h_{ie} + R_l(h_{ie}h_{oe} - h_{fe}h_{re})]} \qquad \ldots(22)$$

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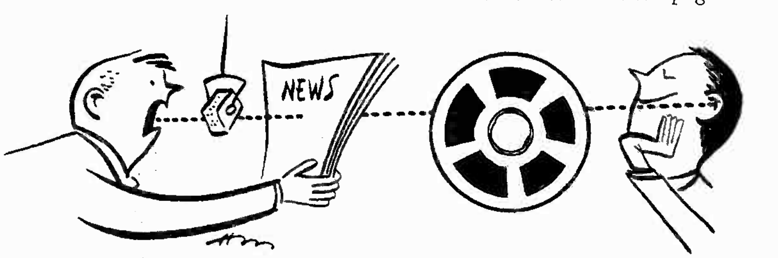

TRANSISTORS

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$$G_t = \frac{4R_g R_l h_{fe}^2}{[(R_g + h_{ie})(1 + R_l h_{oe}) - h_{fe} h_{re} R_l]^2} \quad \ldots(23)$$

Actually not all seven need to be calculated, for, besides the formulas we have for G and G_t, it is generally true for any circuit of the form of Fig. 1 that

$$g_x = \frac{A_i}{R_i + R_g} \quad \ldots(24)$$

$$R_t = -\frac{R_l A_i}{A_v} \quad \ldots(25)$$

Exactly the same sort of analysis can be carried out to find the same relationships for the other two configurations, common-base and common-collector. Since the derivations are exactly the same, there is no need to go through them here.

The Formula Tables

All these formulas can be expressed in terms of the other parameter sets, merely by making use of the transformation equations given previously. When this is done, we are able to find the characteristics of any of the configurations, in terms of any of the parameter sets. Table I this month contains formulas for the grounded-emitter case, in terms of both the common sets of parameters: the h_b set, and the r set.

Tables II and III present the same sort of tabulation as Table I, for the grounded-base and grounded-collector configurations. These two circuits are not used very widely, but the tables are included for completeness.

The three tables give not the exact formulas, but approximate ones, simplified by making the same approximations as were made before. Such approximations are all right for modern junction transistors. But notice that these formulas are valid for *any values* of generator resistance R_g and load R_l.

If we restrict ourselves to customary values of generator and load resistance, that is, a low enough load, and a high

enough generator resistance, the formulas in Table I can be simplified considerably. Whenever R_l is much less than r_d, and when R_g is much greater than $r_b + r_e(1 + \beta)$, the simple formulas in Table IV hold. Quite often in audio amplifiers this is the case, although sometimes it is not. When these conditions are true, then by all means the simplified equations should be used. Even when the conditions are not quite fulfilled, use of the simplified equations can give a quick, rough figure. Later the better formulas in Table I can be compared to see how much the simplified formulas were off.

Tables V and VI were derived for the grounded-base and grounded-collector circuits in the same sort of way. Note the simplifications effected here. The approximations made in the grounded-base case will usually hold in audio work; those made in Table VI quite often will *not*.

This, then, has been the result of our derivation: Tables I and IV give grounded-emitter formulas, the first table good for any values of external resistance, and the second good for certain values only. Similarly Tables II and V, and Tables III and VI, present the same sort of compilation for the rarer common-base and common-collector configurations.

WOODCRAFTER

Continued from page 8

If there is any doubt in your mind as to which it is, ask a builder to look it over. If it is a bearing wall, let the professional handle the cutting into the wall since special support is needed, both during the operation and afterward. If, however, it is a nonbearing partition, then it is something a home handyman can handle easily.

If the built-in can be fitted snugly between existing studs, the job is simplified. Studs are usually spaced 16 in. apart on center. By locating a stud near one end of the proposed opening, it would be a simple matter to measure in multiples of 16 to locate a stud near the opposite end. When the wall is opened, measure the exact distance between the end studs; that is the length to make the built-in. But we're getting ahead of ourselves!

First, how is a stud located? There are a number of ingenious gadgets on the market intended to locate the stud for you, but a good homemade method requires only a hand drill with a small twist drill (1/16 in.) in the chuck. Simply drill into the wall and, if the drill suddenly slips forward after passing through the plaster or wallboard, you will know there is no stud at that point. Move the drill to another spot and try again. If it hits a stud, there will be no slipping and soon tiny chips of wood will appear where the drill is entering the surface of the wall. There should be another stud 16 in. away.

Once the location of the built-in has been decided upon, the outline of the opening should be sketched on the wall. A floor-to-ceiling unit would require but two vertical lines at the extreme ends. If the built-in is not intended to reach the ceiling, a level horizontal line will be required at the top. Never use the floor and ceiling as measuring guides, since settling of the average house throws them off level. Use a straightedge and a spirit level to mark the opening, Fig. 2. (A straightedge can be any perfectly straight length of wood or metal.) If a top line is needed and it is low enough to be reached conveniently, mark the height and, using the straightedge with the level resting on it, set the line and draw it. If the top line is too high to be reached easily, use the straightedge and level to mark a horizontal line at a convenient height. Then, at two points, measure up from this line to the correct height for the top line. Connect these points, using the straightedge for a guide. For the vertical outlines of the installation, mark the

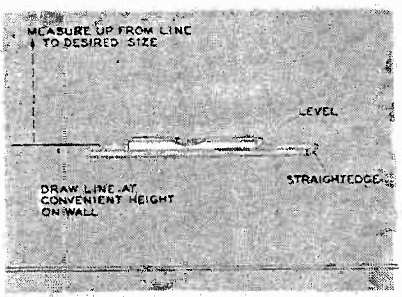

Fig. 2. Using a level and straightedge to set line marking height of built-in.

points at which the lines are to be drawn and set them by using the straightedge and level held vertically instead of horizontally.

With the outline of the operation drawn, we are ready for cutting. We can start the cut by breaking through the plaster with a cold chisel. Then, using a keyhole saw or a portable electric saw of the circular or saber type, follow the line and cut through the plaster and lath. Work with care so as not to disturb the plaster on the wall surface immediately adjoining the location of the built-in. Saw down through the baseboard and shoe molding and pry them off with a wrecking bar. If the rear of the built-in is to project into another room, follow the same process on the wall of that room as on the front wall. If the built-in is to be recessed only the normal depth of the partition, extreme caution is necessary so that no damage is done to the wall on the back.

For the present, let us assume that we are excavating the partition only as deep as the studding will permit. When the cutting of the outline has been completed, the next chore is removing the lath and plaster inside the line. Here again the wrecking bar can be a handy implement. With the wall surfacing gone, we find 2-by-4 studs remaining within the area to be cleared. Removing them requires the extreme caution mentioned a moment ago, for we must remember that the lath on the back wall is also fastened to these studs. Any rough action would cause damage to the wall in the other room. Start by sawing through the studs at intervals; then split the sections length-wise along the grain with a hammer and chisel until they fall

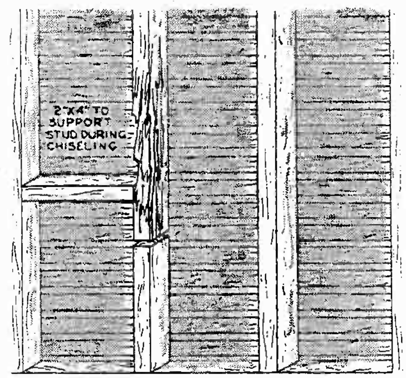

Fig. 3. Stud chiseled from back of wall should be braced firmly as shown here.

free of the lath nails. While chiseling the studs, use a length of two-by-four as a backing wedged between the next solid stud and the one being chiseled (Fig. 3). This will reduce greatly any shock to the rear wall. Once the waste studs have been removed, the exposed ends of the lath nails can be clipped off.

Now the opening is ready to receive the built-in. Accurate measurements from stud to stud and top to bottom should be made and the built-in unit constructed to fit. There is a great difference between the construction of a built-in and a piece of movable furniture. The furniture relies on itself for complete support, therefore special care must be taken in building the joints, legs, and other supporting members. The built-in, however, actually becomes part of the house and gets most of its strength from the floor and walls to which it is attached. Fit it snugly into the frame of the house and you'll have a solid unit with very little gap between the cabinet and the cut edge of the plaster wall. Any space that does remain can be covered with a wood molding.

An excellent backing for a built-in is ¼-inch plywood available in a choice of woods — fir, mahogany, birch, knotty pine, etc. If the back of the unit will not be exposed, it would be more economical to use plywood that is good on one side only. Where design permits, the outside wood trim of the built-in can be the same as that used throughout the remainder of the room.

In cases where the built-in is constructed within a closet or completely recessed within a wall, a new type of

Continued on next page

WOODCRAFTER

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folding door hardware (Fig. 4) permits covering the opening with wood or composition doors with each half center-hinged and, when open, folding against

Courtesy Stanley Hardware, division of the Stanley Works

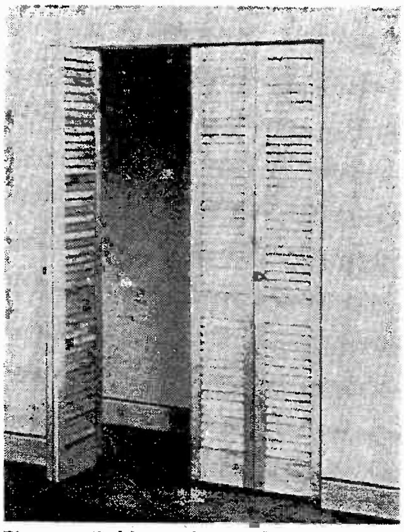

Fig. 4. Folding doors, like this one, can protect and beautify your built-in.

the jamb with a minimum of projection into the room. No bottom guide track is needed.

And now let's sit back and relax a few moments. Cutting into a partition is not the easiest of jobs — but it is very rewarding when you gaze upon the finished work and enjoy its services.

TAPE NEWS

Continued from page 12

to boost them during playback, we would be boosting the tape hiss at the same time. So the obvious place for treble equalization is in the recording circuit.

Now we have a means of recording at low distortion and obtaining linear response. All we have to add to our hypothetical recorder is some means of telling when we're overloading the tape, without actually having to play the whole thing back. The most common type of volume indicator is the neon bulb, which may be used singly as an overload indicator, or in pairs to indicate overload and normal volume levels. Next, there are the so-called "magic eye" fluorescent indicators that register lower-volume signals as well as normal and overload conditions. And finally, there are the VU meters used on professional and semiprofessional recorders. The VU meter is the most effective and versatile of the three types, because it can also serve as a db meter when running certain tests on the equipment.

Fig. 1 shows how we might go about putting all this together to assemble a basic recorder (without playback facilities, for the time being). In this illustration a microphone is feeding several voltage amplifier stages, which in turn deliver the signal to the volume indicator and to the recording equalizer. Since we are acknowledging the existence of the NARTB standard tape reproducing curve, and are using a playback head with a quarter-mil pole-piece gap, we will need treble boost turning over at around 1,500 cps and sweeping up to about 26 db at 15,000 cps. This curve would be difficult to obtain with an RC

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network, so we use a combination of L and C, peaking it at 15,000 cps. Finally, the output from the equalizer is mixed with that from the bias oscillator, and both go on to the recording head.

Following the response curves in Fig. 1, we can get some idea of what has happened so far. When we play that tape back again, the response will droop on both sides of 3,000 cps. The treble, boosted in recording, will come out flat — but equalization is needed to restore the falling-off bass. Fig. 2 shows how, in the average home tape recorder, the same elements as were used for recording are switched around to double as the playback system. In a tape recorder having *separate* heads for recording and playback, there would be two complete amplifying systems built into the recorder which could operate simultaneously. The advantage of being able to listen to the playback from a tape while it is being recorded should not need pointing out.

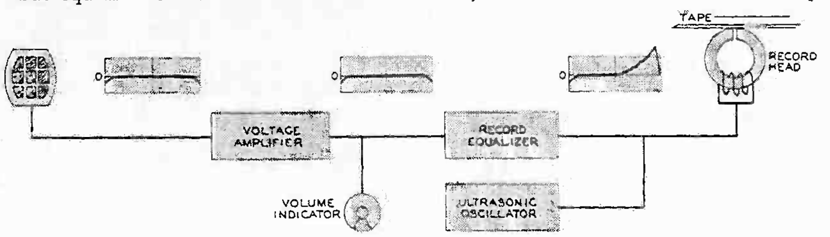

Fig. 1. Signal response at several points in recording circuit of a tape machine.

In Fig. 2, we have replaced the microphone with the tape head, and split the amplifier into two cascaded sections, with the playback bass equalizer (3,000-cps turnover) inserted between. We then have our properly equalized signal, and if we are lucky it will be a pretty good copy of the original signal.

This is not quite *all* there is to tape-recorder design, but it may make us wonder why more manufacturers have not gone into the tape field. It may also tempt us to try making our own recorders, which is quite a worthy project for any electronic genius. But a word of warning: unless you are also a mechanical genius, and have access to a completely equipped tool shop, don't bother to try building a transport mechanism. This is the part of a tape recorder that more leading designers have failed in than any other. Check the speed-regulation specifications on most home recorders, and you'll see what I mean. It's safest to start out with a good tape deck.

One useful project, though, does not require a grasp of the principles of lathesmanship. To my knowledge, there is not yet available a really good, low-distortion stereo preamplifier with accurate NARTB equalization, for use with two high-quality power amplifiers. Enough said!

Tape Discs

I don't believe that this will start a trend, but the first talking book on disc about tape was released recently on the Crest Records label. This 12-inch LP disc, entitled *How to Use Your Tape Recorder*, represents the first time that anyone has used the disc medium to illustrate audibly the right and wrong way to make recordings and it prompts me to ask, rhetorically, why hasn't somebody thought of this before?

Someone *did* think of it before, and has brought out a similar sonically illustrated booklet on a recorded *tape*, but that I have not yet heard. Meanwhile, I'm content to say that *How to Use Your Tape Recorder* is, on the whole, a record that every home recordist should buy, borrow, or copy onto tape. It is necessarily somewhat limited in scope (having to work within the time restrictions of a single disc), but it does manage to cover in considerable detail such matters as frequency response and response testing (with test tones), choice of tape speeds (with examples of music and speech recorded at 15, 7½, and 3¾ ips), correct and incorrect recording level (with examples of each), microphone placement (with sample piano recordings at 1, 4, and 20-foot mike distances), room acoustics (with examples of live and dead rooms), good and bad splices (with examples), head demagnetization (with the sound of a magnetized head), recording off the air (by microphone and direct connection), and duplication and editing of disc recordings.

This adds up to a surprisingly complete picture of the tape recordist's most practical problems, and the whole project is carried out quite successfully. I was disappointed that more space was not devoted to examples of microphone

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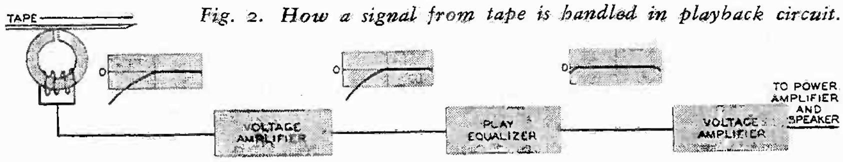
Fig. 2. How a signal from tape is handled in playback circuit.



TAPE NEWS

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technique, but perhaps Crest Records is planning a second volume which will explore that in greater detail. Also, I must say that the audible example of head magnetization was rather extreme, and did not illustrate the description of this condition that preceded it. I suspect someone may have goofed on that one.

All in all, though, a commendable effort and a worth-while addition to any tape recordist's record library.

GROUNDED EAR

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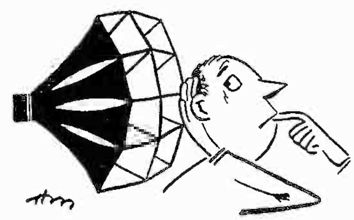
phile discs have filled the bill in the disc field, informs me that he has rather reluctantly been providing a 15-ips tape of selections from the Audiophile library at $18 a tape. These are individually dubbed from the master tapes on Mr. Nunn's highly refined tape recorders at 15 ips, and they do not have the faults of tapes produced on high-speed duplicators. The tapes are made to order with the customer's choice of material from the Audiophile disc library. This is strictly an accommodation, and Mr. Nunn hopes that the price will discourage all but those who have a use for the tapes which will further the acceptance of tape reproduction. For details write E. D. Nunn, Audiophile Records, Saukville, Wisconsin.



READERS' FORUM

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which I still have not done. But, as is, performance is equal to that of machines in the $300 to $500 price range.

For those who may be disturbed about the high cost of the hysteresis-synchronous motor, it is possible for a group to get together and place an order for a number of motors at a reduction in price. Several phono turntables on the market are offered with a choice of hysteresis-synchronous or induction motors in the same models, and in every case the price *difference* is about $45.

All in all, I feel that the venture was a complete success, and I now have a better tape recorder better than any I could have afforded to buy outright. My thanks to Mr. Geraci and AUDIOCRAFT.

Edwin D. Bates
Elyria, Ohio

ADVERTISING INDEX