audiocraft

DECEMBER THE HOW-TO-DO-IT MAGAZINE OF HOME SOUND REPRODUCTION 35 CENTS

THIS MONTH

CHASSIS LAYOUT AND WIRING

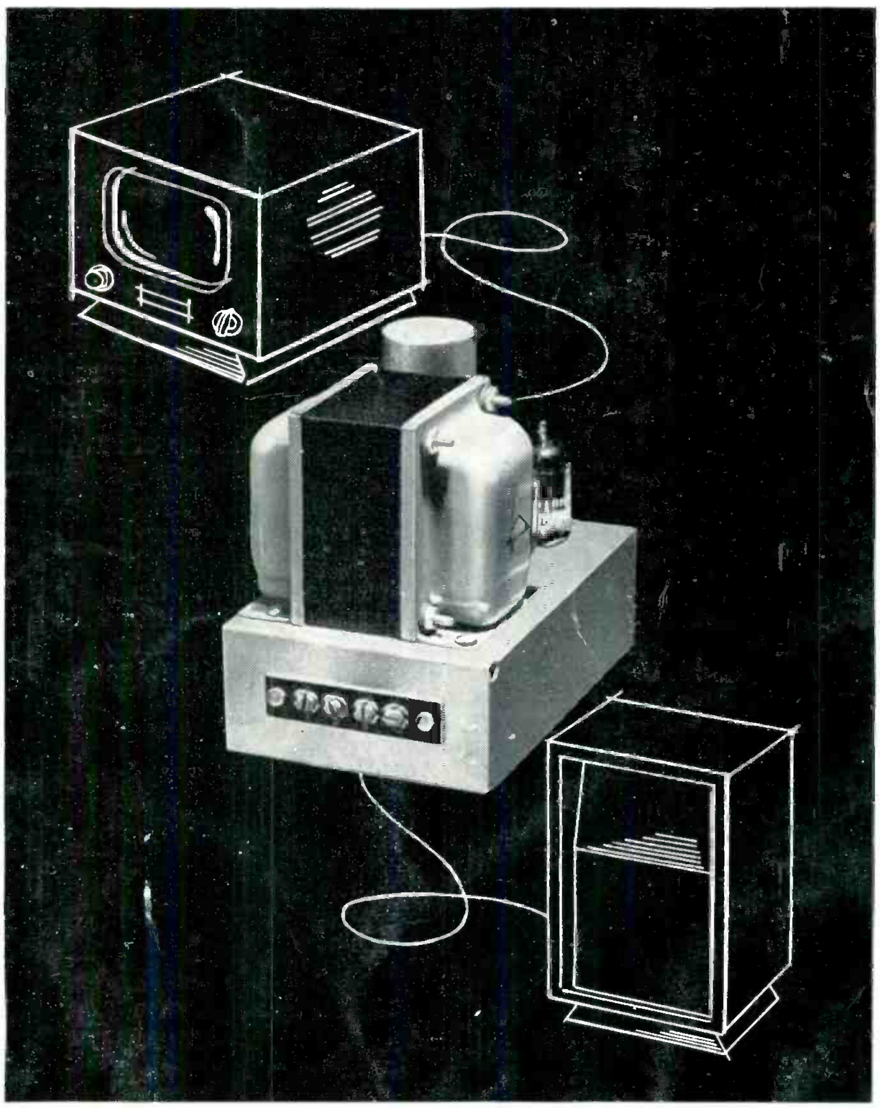
How to lay out, scribe, and cut chassis for home-built preamps, amplifiers, and similar units.

THREE-CHANNEL MICROPHONE MIXER

A truly high-quality mixer that you can build yourself. Noise and distortion are both negligible.

A TURNTABLE WITH STRING DRIVE

How to quiet your present turntable—How to build a deluxe turntable at relatively low cost.



TV CAN SOUND BETTER

Here is a small and inexpensive high-quality amplifier to replace the sound output stage in a TV set or commercial radio-phonograph. Easy to build, too, with the pictorial diagram furnished.

CABINART '56 – HI FI CABINET KITS

... all you need is a screwdriver!

Machined wood pieces key exactly to each other! Pre-shaped, pre-drilled, pre-engineered and not a scrap of sawdust left over!

THE REBEL K-3

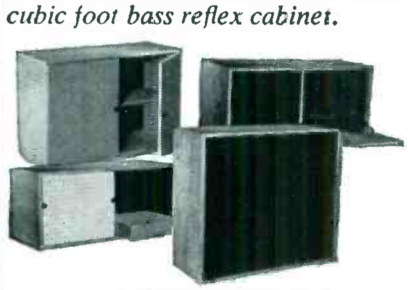
Identical acoustically with the KR-3, first and largest of the Klipsch-designed Rebel series of corner folded horns. Using the mirror images of room walls at a corner, the K-3 extends bass down nearly to 30 cycles! Two companion Rebel kits are more economical but only in price and size.

K-3 . . . $54.00

Prices slightly higher west and south

THE "ST" SERIES

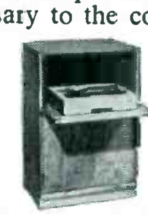
Wall storage units . . . ten basic Cabinart designs in kitform or assembled, ready to finish. Each is tailored to the needs of hi-fi installations. The Series includes a *nine cubic foot bass reflex cabinet.*



Prices from $21.00 to $36.00

FURNITURE KITS

The new 27K and 28K typify Cabinart kit utility and economy. Both point up the unique design functions necessary to the correct hi-fi installation.



Equipment Cabinet $51.00

Bass Reflex $39.00

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kits by cabinart ... *the pioneers in radio furniture for high fidelity.*

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WRITE

1. Equipment Storage
2. Equipment/Speaker Enclosure Kits
3. The Rebel Speakers
4. Wall Storage Cabinets
5. Accessories By Cabinart

* Trademark

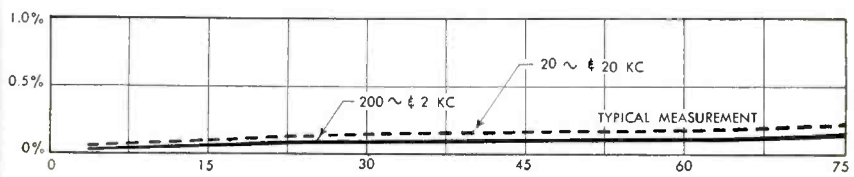
Here's Proof of McIntosh superior performance!

For *clean amplification, low distortion* and *abundant power* no other amplifier compares with the McIntosh—long the standard of high fidelity excellence. The fundamentally-different McIntosh circuit delivers amplification within 0.4% of theoretical perfection. Nothing is added to or taken from the input signal. The result: a realism, clarity and listening quality without "fatigue" caused when tones are lost, distorted or intermixed within an amplifier. There are more *plus* values with the McIntosh. Hear it at your dealer's.

Distortion: 1/3% Harmonic and 1/2% IM, even at full rated output, from 20 to 20,000 c.p.s. *Power:* 30 watts continuous, 60 watts peak (for Model MC-30); 60 watts continuous, 120 watts peak (for Model MC-60).
Frequency Response: 20 to 20,000 c.p.s. ±0.1 db at full rated output. 10 to 100,000 c.p.s. ±1.0 db at one-half rated output. High efficiency of the McIntosh circuit means longer life, less heat dissipation and less power consumption for greater output.

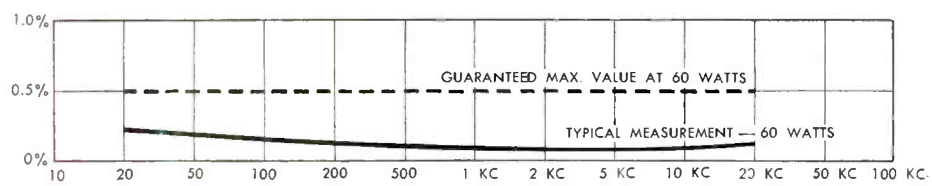
The exclusive, patented McIntosh Bi-filar Circuit, first with Unity Coupling, is inherently superior by design, laboratory measurement and listening test. Wave-form distortion, due to switching transients between each half of the class "B" amplifier is eliminated at all frequencies. The result is the purest amplification possible.

DISTORTION VS. POWER OUTPUT

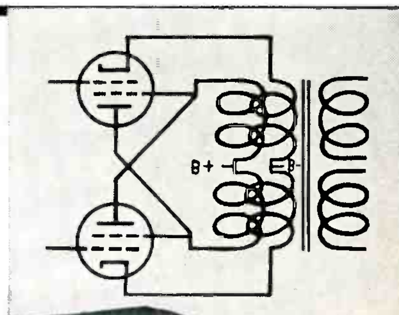


POWER OUTPUT IN RMS WATTS

SINGLE FREQUENCY HARMONIC DISTORTION



FREQUENCY IN CYCLES



Write for Free Booklet "Lost Instruments" and complete specifications on McIntosh amplifiers and control units.

MC-60 $198.50

McIntosh

LABORATORY INC.

325 WATER STREET • BINGHAMTON, N. Y.

EXPORT DIVISION: 25 WARREN ST., NEW YORK 7, N. Y. CABLE: SIMONTRICE, NEW YORK

by

h.h. Scott

Revolutionary New Turntable Has Everything

710-A Stroboscopic Turntable, $125.00*

New acoustic filtering keeps out ALL interference

■ New turntable design principle, acoustic filtering, prevents speaker, building and motor vibrations from ever reaching the turntable. This frees record playing from distortion found in conventional systems.

■ Center-gear drive, with torsional filtering, eliminates "garbling" of high frequencies which results from the flutter inherent in rim drive.

■ Separate vernier control of each speed allows super-exact pitch adjustment. Convenient pushbutton selection of 33⅓, 45 and 78 rpm speeds.

■ Optical stroboscope for extremely precise speed settings, even while record is playing.

■ Built-in vibration isolation and pickup arm mounting system simplify installation.

TECHNICAL SPECIFICATIONS

Rumble more than 60 db below recording level — wow and flutter less than 0.1% — built-in slip-clutch permits cueing — heavy non-magnetic cast aluminum turntable — heavy-duty special induction motor with dynamically balanced rotor and extremely low hum field — pickup arm mounting board furnished with turntable — dimensions: 16⅞" x 14½" x 7⅛" — accessory mahogany base $14.95*
*Slightly higher west of Rockies.

All-In-One AM-FM with Equalizer Preamplifier

by

h.h. Scott

331 AM-FM (Binaural) Tuner $189.95*

The perfect answer where space is at a premium

■ Includes complete equalizer-preamplifier with Bass, Treble and Loudness controls, plus four-position record compensator.

■ Same sensational AM, FM, and binaural performance as in 330 tuner described above.

■ Special provisions for playback of pre-recorded tape through your music system.

■ New two-speed planetary-drive tuning: high speed for instant station choice, slow speed for precise tuning to weak stations.

■ New chassis design of the 331 and 330 makes custom installation very simple. Beautiful accessory case available for using tuner on table top or shelf.

TECHNICAL SPECIFICATIONS

FM and AM sections same as 330, above — selector switch for two high level inputs, four equalization curves (RIAA-NARTB-Ortho., Orig. AES, Orig. Col, EUR 78), NARTB tape playback, FM, AM wide range, AM normal, AM distance — bass and treble controls — two magnetic pickup inputs — recommended for use with any H. H. Scott power amplifier — beautiful accessory case $9.95
*Slightly higher west of Rockies.

Write for FREE BOOKLET giving complete details on entire H. H. Scott line.

h.h. Scott

385 PUTNAM AVE. • CAMBRIDGE 39, MASS.

December 1955

Volume 1 Number 2

audiocraft

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

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

John J. Huff lives in Stockton, Calif. He says this article is his first literary effort. We hope it won't be his last, because it's an unusually excellent beginning.

Edgar M. Villchur is president of Acoustic Research, Inc., and a man of many parts — educator, author, and audio design engineer, to name a few. He conceived and developed the acoustic suspension speaker system, one of the few really original loudspeaker ideas in many years.

Glen Southworth has been writing outstanding articles on audio for several years. With this background the Army, when it captured him recently, should logically have been made a paratrooper or something like that; instead, somebody goofed and he's assigned to television work at Camp Gordon, Ga.

S. R. Williams, who writes candidly about turntables and changers, is another Californian — Los Angeles, to be specific. We'll have more material from him.

Ernest B. Schoedsack completes our California representation. Retired and living in Santa Monica, much of his time now is devoted to the search for perfection in audio.

John Hoke, in Washington, D. C., manages technical operations for the motion picture and photographic laboratories of the AAA. He collects and tapes old records, as you might guess, in his spare time.

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The Grounded Ear *by Joseph Marshall*

Cascode Preamplifiers

The cascode amplifier has found application as an input stage in microphone preamplifiers and has recently appeared in a few commercial phono preamplifiers and control units. The circuit has the same advantages for audio use as for RF use: it has the most favorable ratio of signal to circuit noise of any input circuit so far developed. In addition, it has low distortion and is resistant to overloading. Offhand it would seem that it ought to find wide application as a phono preamplifier. There are several reasons, however, why it has not and probably will not displace completely the conventional cascaded circuit employing such tubes as the 12AY7, 12AT7, and 12AX7.

In the first place, whether the tube used is one of the above twin triodes or an RF twin triode like the 6BK7 or 6BQ7, a single cascode stage will not provide enough gain to do the entire job of equalizing and amplifying. With 150 volts on the plate (about the maximum that can be delivered to an input stage if good decoupling and filtering are to be achieved), these tubes will furnish gains ranging from 75 to 125. But the RIAA curve needs 30 db equalization and some other curves need even more. Moreover, the preamp needs an additional gain of anywhere from 26 to 35 db to deliver an adequate output voltage with low-level cartridges. Clearly, then, a cascode stage must be followed by another stage of amplification: that means another half-section of a 12AT7 or 12AX7. In an elaborate control unit some use could be found for the other half-section. But since a single 12AT7 or 12AX7 can do the whole job alone adequately, the cascode stage is at a disadvantage and the complication would have to be justified by a considerable improvement in performance.

And there's the rub. It is difficult to obtain any significant improvement. This seeming paradox is the result of one of those hard-to-face facts of life. The fact is that it is much more difficult to lower the noise threshold of a wide-band audio stage than of a tuned RF stage, even a very broad-band one, and the best of today's cascaded designs already approach the irreducible threshold so closely that further improvement is by no means as significant as one would expect.

The cascode stage is superior principally in improving the ratio of input signal to *tube noise.* But tube noise in audio stages is a factor only when other circuit noises have been all but eliminated. The situation is much more unfavorable in the case of a phono preamp than in the case of a mike preamp, because the equalizer has the effect of reducing the tube noise by treble rolloff and of increasing the hum by bass boost. Even when DC is used on the filaments, some pick-up of hum by induction is practically inevitable. The modern household has a surprisingly large AC field, modulated by various noises, because it is within the gigantic loop of the house wiring through which a heavy current flows: it takes extremely effective shielding to minimize pickup of this hum and noise.

Assuming that hum has been reduced to an insignificant extent and DC is used on the filaments, a cascode stage can give an improvement of 6 to 10 db over a 12AT7, but not nearly as much over a 12AY7. When AC is used on the filaments I doubt that any real improvement is possible because the residual hum will be the largest fraction of residual noise and even a considerable reduction in tube noise will have little effect on total noise.

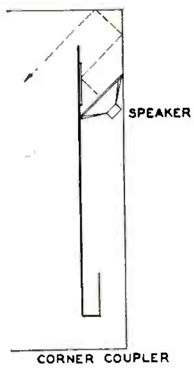


Fig. 1. A speaker system that takes very little floor space in a corner.

An improvement of 6 to 10 db is, of course, not to be sneered at and it may be quite significant if: 1) the turntable has a very low noise level; 2) the pickup contributes little or no hum; and 3) the recordings themselves are free of noise. But in an average installation it is very likely either to go entirely unnoticed or merely to unmask other and equally annoying noises.

This is not to belittle the cascode. It will be invaluable for such low-level pickups as the Fairchild, Ferranti, and Angel. (That is why Fairchild use it in their preamp.) It will also be worthwhile in top-notch installations where other noise has been reduced to a minimum. And for those to whom the best is not quite good enough, the added complication and cost will not matter. But these facts explain why the cascode has not swept the field of commercial equipment despite the fine advertising value its reputation gives it.

The Corner Coupler

I suppose that if a Gallup poll were taken of audiophiles, it would be discovered that what most of them want most is a speaker system with a response to 20 cycles, small enough to put in the third-floor back of a Greenwich Village brownstone, and cheap enough to leave at least a few pennies of change from a hundred-dollar bill. Antony Doschek, an excellent violinist (Pittsburgh and Chicago Symphonies, WQXR, etc.) and a competent engineer as well (Tribo-Electric Metal Sorter) has designed and is selling* an item which fills this bill much better than I supposed was possible until I heard it.

The Corner Coupler is a box 6 ft. tall, 14 in. wide, and 12 in. deep which stands upright in a corner and occupies about as little useful floor or wall space as a speaker enclosure can. It combines the principles of the tuned open pipe, bass-reflex, corner horn, and mutual coupling of radiators with images to achieve a response which is audibly quite flat to 20 cycles. The bass produced should be capable of satisfying even those whose yearning for big bass is on the pathological side. Furthermore, it delivers this bass even at extremely low volume levels. Doschek himself has some reservations about the quality of the bass, and it is not as clean and well defined, to my ears, as a perfect speaker system should be — but then, this is not yet the millenium when $100 will buy perfection. The several people I know who own it are entirely happy with it and with the open-mouthed envy its response produces in owners of systems with a more restricted range and lower bass efficiency.

Fig. 1 gives a rough idea of the system when it is installed in a corner. The speaker is an 8-inch Stentorian (or, in some models, a Goodmans Axiette) and is mounted close to the top of the enclosure so that the bass frequencies from the front radiation work into the expanding horn provided by the corner, while the higher frequencies are reflected from a pane of glass to the walls and ceiling and dispersed throughout the

*Pro-Plane Sound Systems, Inc., 1101 Western Ave., Pittsburgh 33, Pa.

room. This dispersion of highs eliminates the directional effect, but at the cost of some loss — partly by absorption, but largely because, when a given sound is spread over a wide arc, obviously less is sent to any given spot than could be done by directing it in a narrow beam. This loss can be corrected with a tweeter for those who mind it. But the effect without a tweeter approximates the natural sound of dispersed highs in an auditorium.

The rear of the speaker works into a chamber with a port at the bottom. You can call this an open column, or a bass-reflex, or see similarity to an air coupler or an acoustic labyrinth (without the labyrinth). The column is carefully and critically tuned to make up for speaker slope, and is damped by critically proportioned and positioned pads of glass wool. The port works into the corner at the floor and profits from the corner position and the horn-like expansion.

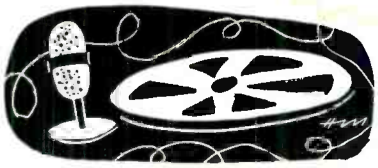
Doschek has an interesting theory about speaker systems and the corner coupler reflects that theory. He points out that bass tones (as well as all musical tones) are generated by highly individual and dissimilar instruments. Drums are diaphragms, the pedal tones of an organ are produced by tuned pipes, the bass viol is a Helmholtz resonator, and the brass basses are horns. He says that no single type of speaker can be expected to do them all justice: if it reproduces the harmonic structure accurately it may not be able to reproduce the attack and decay characteristics, etc. Certainly many others also believe (for example) that horns are rather poor on drums, and diaphragm-direct-radiators are poor with organ bass. At any rate, he designed the corner coupler primarily to provide the needed variety of generators with a single driving element. He obtained the extraordinary extension of range downward as a bonus which, though by no means accidental, was not predictable on mere theory alone.

Moving-Needle Pickup

In the search for a phonograph pickup of ideal compliance, many obscure electrical principles have been resurrected and put to practical use. The variable reluctance pickup, which revolutionized record production, is a case in point: the principle had previously been of mere academic interest.

Pickups are the inverse of motors or galvanometers: whereas motors convert an electric current into mechanical motion, pickups convert mechanical motion into electric current. Moreover, the compliance required of pickups is on the order of that required of galvanometers. So it was natural that designers should go to electrical instruments for models of pickups. We have had some excellent moving-coil pickups, including a faith-

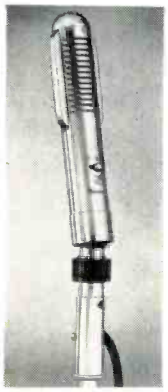
Continued on page 43



AUDIONEWS

MINIATURE FEN-TONE B & O-50 RIBBON MIKE

The Danish-made B&O-50 is the third Fentone Blue Ribbon mike introduced this year. The microphone has a 3-way switch with "Close Talk" (T), "Music" (M), and "Off" (O) positions, and a ball swivel mounting. The snap action stand connector with standard ⅝ in. by 27 threads is attachable to standard microphone stands and booms. The microphone is shipped with 20 ft. of shielded, balanced, 3-conductor cable. A desk stand is also available at additional cost.



Fen-Tone B&O-50 Ribbon Mike.

A partial list of manufacturer's specifications follows:

- Secondary Impedance: 50Ω
- Frequency Range: 30 to 15,000 cps ±2½ db
- Sensitivity at 1,000 cps: "T" position —59 db; "M" Position —55 db
- Size: 7¾ in. by 1 3/16 in.
- Weight: Less than 15 oz.
- Non-corrosive duralumin ribbon: 0.0001 in. thick with 0.0013 gram mass.

Price of the microphone is $48.95.

For further information about the B&O-50 microphone, write to Fenton Company, 15 Moore St., New York 4, N. Y.

BOGEN SHORTWAVE HIGH FIDELITY RECEIVER

A new high fidelity AM receiver for use on shortwave bands and the broadcast band has been introduced by the David Bogen Co., Inc.

Known as the RR29, this 11-tube, superheterodyne, 6-band receiver is available in chassis form or, as model RR29W, in a mahogany veneer cabinet complete with 2 loudspeakers. It is designed to operate on 110 volts direct current and on 110, 150, and 220 volts, 50-60 cycle alternating current. Shortwave coverage from 4.7 to 18.1 megacycles is provided in 5 bands.

The RR29 has a response stated as down 3 db at 40 and 4,000 cycles and a power output of 2 watts at 2% distortion. Band width is 8 kilocycles for 3 db and 20 kilocycles for 56 db attenuation. Sensitivity is rated at 1 microvolt and 30 microvolts for 6 db and 40 db signal-to-noise ratios respectively. A noise clipper circuit reduces noise in proportion to the percentage of the modulation of the incoming signal. Separate inputs for phonograph and auxiliary source permit playing of both records and tape.

The RR29 is priced at $165.00 and the RR29W at $199.50. If further information is desired, write to David Bogen Co., Inc., 29 Ninth Ave., New York 14, N. Y.

NEW JAMES TAPE RECORDER

James Instrument Laboratory is now producing a new tape recorder, the V-12.

The unit features a 3-motor mechanism with a 2-speed, hysteresis synchronous motor for drive. The machine will handle reels up to the 10½-inch NARTB size without adapters and employs electro-dynamic braking for smooth and safe stoppage. All tape motion is controlled by electrically interlocked push-buttons with automatic stop in case of tape breakage or run-out. The tape guides automatically release the tape from contact with the heads during fast-forward and rewind operations, and accidental erasure is prevented by an electrically interlocked "record" button. Tape speeds are 7½ and 15 ips, or 3¾ and 7½ ips.

The recording amplifier has separate circuits for recording and playing back, and standard NARTB playback equalization. The unit has a 10-watt push-pull monitoring and playback amplifier, cathode follower output for external amplifier, mixer for 2 inputs, and a 4-inch VU meter. Over-all frequency response is said to be 40 to 15,000 cps at the 7½-ips speed.

James V-12 Tape Recorder.

For further information about the V-12 tape recorder, write to James Instrument Laboratory, 9110 South 52nd Court, Oak Lawn, Ill.

EARPHONE AGGREGATE BOX

A new attachment known as the "Earphone Aggregate Box" has been developed by the Audio-Master Corp., 17 East 45th St., New York 17, N. Y. This unit, which distributes sound to individual headsets, is housed in a compact metal case and covered with a Bakelite top. A 10-foot extension cord completes the unit.

The implementation of this attachment makes possible the use of as many as 8 headsets for individual earphone listening: it can be used in conjunction with any record or transcription player, tape recorder, or radio receiver having a detachable loudspeaker or a special jack for earphone use.

Earphone Aggregate Box.

Fisher 80-AZ Amplifier

80-T FM-AM Tuner/Control Unit.

TWO NEW FISHER FM-AM TUNERS, NEW AMPLIFIER

Fisher Radio Corporation has announced 2 new FM-AM tuners, Models 80-R and 80-T. Both units feature 2 meters for accurate tuning on FM and AM.

The Fisher Model 80-R is designed for use with an external control chassis. The Model 80-T has the same characteristics as the 80-R, but features, in addition, complete control facilities, including a tape-head playback preamplifier equalized to the NARTB characteristic.

The Model 80-R is priced at $169.50: the Model 80-T is priced at $199.50. Mahogany or blond cabinets for both units are available at $17.95 each.

In addition to the 2 tuners, Fisher has also announced a new 30-watt amplifier, Model 80-AZ featuring "Z-Matic" circuitry and "Powerscope", a peak power indicator calibrated in watts to show instantly the peak load on the speaker system.

The Model 80-AZ is priced at $99.50. For further information about the 80-AZ amplifier and the 80-R and 80-T tuners, write to Fisher Radio Corporation, 21-21 44th Drive, Long Island City 1, N. Y.

NEW MALE PRINTED CIRCUIT CONNECTOR

A new series of male printed circuit connectors has been announced by the Circon Component Company. The new connectors are said to be a decided improvement over previous models.

The manufacturer states that the new connectors provide higher insulation resistance and humidity performance due to the use of a new phenolic base material. A sustained, near-perfect contact is maintained with the new shot-burnished gold deposition process employed in the production of the contact surfaces. Interface bond between the contact and the body of the connector is such as to eliminate possibility of damage to the contact bond due to careless application of the wires to contacts. Increase in width of extremity contacts provides sound cabling and relief of strain at these critical points. A vinyl insulating hood provides a built-in mold for potting of the connectors if the user desires.

For complete information about this series of connectors write to Circon Component Company, 17544 Raymer St., Northridge, Calif.

> $5.00 FOR AUDIO AIDS
>
> *We'll pay $5.00 for Audio Aids sent to us by readers and used in the magazine. See page 43 for details.*

Appropriate Settings: No. 1

OCCASIONALLY we receive photographs of equipment, like the one shown here, in which interest centers not entirely on the equipment. We thought this model deserved something in the way of special attention and so we've given her a little extra space. Due to an unfortunate oversight on the part of the Promotion Department of the Ampro Corporation, no information about the young lady — not even her name — was given in the press release accompanying the photograph.

The tape recorder (right center) is the new Ampro "Consolette", Model 757B. The Consolette comes in 2 models, the 757B which has a blond finish, and the 757M in red mahogany. These new Ampro models feature a 2-speaker system, an amplifier bypass for high fidelity hookups, piano key controls, automatic selection locator, and an electron eye recording level indicator. The machine records and plays back at 7½ ips or 3¾ ips. Frequency response is stated to be from 40 to 12,500 cycles.

List price for either the 757B or 757M is $279.95. Matching stands are available for $17.50. Recorder-Radio combinations of these models are available at $34.50 extra.

Further information about the Ampro 757 may be obtained from the Ampro Corporation, 2835 North Western Ave., Chicago 18, Ill.

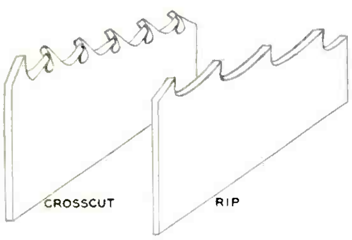
Tips for the Woodcrafter

by George Bowe

IN my youth there was no mistaking when Christmas was just around the corner. The first official notice did not come from the lovely carols nor the dazzling store windows but, rather, the telltale decorations on Grandfather's fingers. No bells or ornaments did Grandpa sport — his hands were more reminiscent of a visit to the medicine chest for, along about this time of year, his fingers began wearing homemade bandages. Nothing serious, mind you, just small wrappings of gauze and adhesive tape to cover little nicks and scrapes acquired at the workbench in the cellar. Regularly each year about 6 weeks before Christmas an air of great secrecy began to cover Grandpa's subterranean activities, but the bandaged fingers told us what he was about. We knew that underneath the glowing tree on that glorious morn each of us would find a thrilling sample of his workmanship — a jumping jack or a pirate sword, a cradle or a scooter.

Grandpa loved to work with wood, and his creations were attractive and substantial. Since those days I have often wondered why he suffered so many minor cuts and bruises. Was it because his woodworking activity was almost entirely limited to one period each year — the Santa Claus season? Or was it because he had never been taught the correct use of tools? I think that Grandpa would have benefited considerably if someone had passed along to him the rudiments of proper handling of the basic tools. The right way is not only the safer way, but the route to better workmanship.

The saw is one of the most frequently used and misused tools. Let's consider some of the things necessary to make the saw deliver the clean, straight edge we want. There are many different shapes, sizes, and types of saws, each designed for a particular job. Of the hand saw varieties, the crosscut saw is most commonly used and, as its name implies, does its work across the grain of the wood. The teeth are filed to a sharp outside edge and point, being set alternately to the left and right to allow proper clearance for the blade while cutting. The fineness or coarseness of the cut is determined by the number of teeth per inch. Keep the saw sharp and clean. Filing a saw is a task for an expert: don't risk ruining the saw by doing it yourself. Have a professional do the sharpening — in some localities the fee is only a dollar. Coating the blade with household liquid wax will preserve the finish and protect it against rust.



Tooth formation of crosscut and rip saws.

Here are a few things to keep in mind when using a crosscut saw:

1) Keep the best side of the wood facing you, since the saw leaves a rough cut on the underside.

2) For accuracy, use a sharp pencil to mark the cutting line. With a blunt pencil you may be off as much as 1/8 in.

3) Grasp the saw firmly, with the thumb and index finger extending along the sides of the handle. The outstretched finger helps to guide the stroke and avoid cutting on a slant.

4) Saw along the outside of the cutting line, not on the line, or you'll find that the saw (because of its thickness) has cut the wood shorter than desired.

5) As you begin, draw the saw slowly up toward you several times at the point where the cut is to start. With the thumb of the left hand in a bent position guide the upward stroke with the knuckle. Bending the thumb avoids the possibility of a bad cut should the saw jump. Never start with a downward stroke, or the saw is likely to jump position, cutting the wood in the wrong place as well as injuring your thumb.

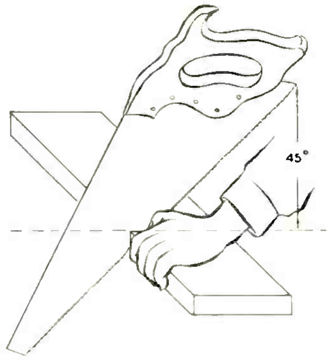
6) Keep the saw at a 45° angle to get maximum cutting efficiency from the teeth.

7) Use *all* the saw when cutting. Take long, smooth strokes and let the saw do the cutting; use pressure only on the downstrokes. Don't twist the body — pivot the strokes at the shoulder and elbow.

8) For a square cut the blade must not tilt to either side. This can be checked by holding a try square against the side of the blade and the face of the wood.

9) When you approach the end of the cut, use light strokes and, with your left hand, support the end of the wood being sawed off. This will avoid splintering the end of an otherwise good cut.

Occasionally a situation will demand that a length of wood be cut with the grain rather than across the grain. The ripsaw is designed for this purpose; it has teeth filed straight across, which act like a gang of chisels. The saw is held in the same manner as the crosscut but at a different angle, about 60° between the saw and the wood. It is even more important in ripsawing to take long, easy strokes in order to assure adherance to the cutting line. In fact, except for the difference in angle, all rules that apply to crosscut saw operation apply also to the ripsaw. In making a long cut, if the wood tends to close in and bind on the saw, simply drop an ordinary table knife, point end first, into the cut. This will act as a wedge to keep the wood from closing in completely as you saw, and the handle of the knife will prevent it from falling through to the floor. As your sawing progresses, move the knife along in the cut.



How to start a cut with the crosscut saw.

There are other saws, such as the back saw and compass saw, whose special talents are often utilized in hi-fi woodworking. Their use can best be explained when we discuss construction details in a subsequent issue. For the moment let's turn our attention to a tool we all take for granted and believe we can use successfully: the hammer. Check yourself against this list of hammer hints and judge for yourself whether or not you should go to the head of the class:

Continued on page 45

LECTRONICS

CONTINUALLY MOVING AHEAD 'TOWARDS PERFECTION IN SOUND'

All three of the developments featured this month have been exhaustively tested and approved in the LECTRONICS sound laboratories, and are personally recommended to you by Irving M. Fried. Request literature, or order with confidence by mail.

by ESL—

Professional Series — Arm & Cartridge

A Truly Professional ARM and Cartridge . . .

The connoisseur of fine music seeking the perfection of reproduction for the most critical ear will welcome this product of Danish craftsmanship. Utilizes the new *moving coil* design with extremely low mass. Features high compliance, new smoothness, very low surface noise, and an absolute minimum of record wear.

ESL-310 Arm (15¾") with diamond cartridge **$106.50**
Diamond: ESL-C1 (.001")or ESL-C3 (.003") **35.95**

The Mark II DYNA-KIT

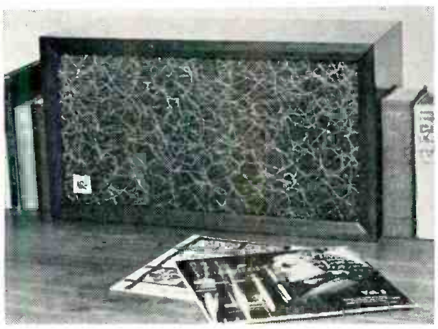
Designed By David Hafler

50 Watt Amplifier You Can Build Yourself . . .

New low-price, "bug" free kit — latest design of Mr. David Hafler, inventor of the ultra-linear circuit which revolutionized amplifier performance. New simplicity of design assures:— 1. Ease of construction and superb performance; 2. Consistent performance with no deterioration of components; 3. Full 50 watts at very little more than the cost of older design 25 watt units. Furnished with complete detailed instructions and diagrams.

Specifications: Power — 50 Watts, continuous at less than 1% I.M.
Power Curve — ±1 db — 20c — 20 kc.
Frequency Response — .5 db — 6 c to 60 kc; Size 9" x 9" x 6⅝" h.
DYNA-KIT (Mark II) 50 W. Amplifier Kit, complete—**$69.75**

By Acoustic Research



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A Revolution in Engineered Acoustics

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Full Range System—Mahogany (AR-1) **$185.00**
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LECTRONICS

CITY LINE CENTER
PHILADELPHIA 31, PA.

SOUND SERVICING

by Irving M. Fried

About Record Changers

In last month's discussion of changers I indicated that there would be a later treatment of problems unique to the assorted gears, levers, springs, cams, etc., that make the changer "change". However, on reconsideration, it becomes apparent that the problems of each changer are so unique unto that particular brand that it is impossible to give general ideas on changer repair beyond those already discussed.

If you have problems underneath your changer, and if your instruction booklet doesn't furnish enough information, it is heartily recommended that you obtain a copy of the applicable Sams *Photofacts* manual, available at any good parts supplier. Common troubles are discussed, and the parts are clearly laid out; the mechanically inclined person will be able to use a *Photofacts* manual to good advantage.

External Noises

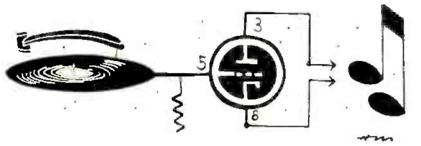
Another series of problems all too common in high fidelity equipment installations is the presence of spurious noises in the reproduced sound — hisses, hums, bongs (microphonics), roars, and thumps. You have all had the experience of wondering where many of these noises originate and, more important, how they can be cleared out. Perhaps you have already had the dismaying experience of buying a new preamplifier for the one you thought was creating the noises, and then finding no improvement.

Actually, high fidelity equipment does *not* tend to be noisier than commercial "console" units. But when you purchased your hi-fi equipment, you were indicating that you were critical, and intended to make exacting demands in reproduction that were beyond the capabilities of commercial units. Most commercial sets have various noises, particularly hum, which "let you know that it's turned on". But no one wants noises in his custom set to let him know; he has pilot lights for that — and the same noise that is not bothersome on the console is disastrous, relatively speaking, on a better reproducer. No one wants to hear a steady background hum on a live FM program, or assorted noises along with his new LP's. But how can you avoid noise?

There are three principal sources, or causes, of extra noise. It may be created within your amplifier or associated electronic units, such as tape preamplifiers and tuners; it may be fed into them; or it may come about through less than satisfactory selection, interconnection, or placement of various component parts. As in any cure, the first and most important step must be indentification of the cause.

The rest of this article will treat only one small part of the problem; namely hum effects apparent in phonograph reproduction, which are not faults of the preamplifier or power amplifier.

Let us suppose that you have a bothersome hum, or some obscure noise, only when you play records. Chances are just as great that you are feeding the noise into your amplifier as that the amplifier is acting up. Just to be sure, try the following: Turn down your volume control. Pull out the phono plug. Then, gradually and carefully, turn up the volume control to its normal position with your input selector switch turned to the phono position (you must be careful, because certain preamplifiers will tend to oscillate or "take off" without the comparatively low-impedance cartridge attached). If the noise is no



longer present, you can be sure now that you have been feeding it into the phono input either from your cartridge, the input wiring or, strange and sad to say, from other wiring in your system. The following checks, singly or together, may help you find your own particular problem.

1) Check the DC resistance of your cartridge on an ohmmeter to see whether or not it is reasonably close to the specified value. Several manufacturers of moving coil cartridges have had partially open coils from time to time — enough resistance remains to give you sound, but with strong hum. Obviously, if your coil is open, replacement is indicated. If you can't get access to an ohmmeter, try inserting and removing the cartridge from the circuit. If there is *no* difference in hum level when the cartridge is attached or if the hum level goes up, you probably have an open coil condition; if it goes down, probably not.

2) Move the arm (with cartridge inserted) slowly across the turntable. If you can hear hum increasing or decreasing, you are getting inductive hum pickup from a motor (probably the turntable) or nearby power transformer. To check the turntable motor: turn it on and move the arm back and forth; then turn the motor off and move the arm again. If the hum disappears with the motor off, it is obviously the motor whose hum field is creeping into your cartridge. The remedy involves changing your cartridge to one of the types with inherently less hum pickup, or trying to rig up shielding between the motor and the cartridge, or replacing your changer or turntable with a type having better magnetic shielding or smaller hum field. For instance, the variable reluctance cartridges such as Pickering, GE, and Audak can sometimes pick up a great deal of hum from inexpensive changer motors. The more satisfactory remedy in the long run, of course, is likely to be a better turntable.

If the hum *did not* disappear with the turntable motor switched off, even though motion across the turntable varied its volume, the cartridge is probably picking up hum from a nearby power transformer. Reorient your components experimentally, turning them at various angles to one another and changing relative positions, and you may find orientations at which this kind of trouble is alleviated.

3) Check the cable from your pickup arm to the phono input plug. For instance, judicious shielding around the terminal strip of a changer can often cut hum amazingly. Try covering the terminal lugs with insulating tape; then use aluminum foil over the entire terminal strip, grounding the aluminum foil to the outside shield of the cable.

4) Check connections to the cartridge. Often, tightening the slip-on clips will make a better connection, reducing hum. Make sure, too, if your changer has plug-in heads, that the prongs on the head fit tightly into the arm connections; if your cartridge slides into a transcription arm, that the spring clips of the arm tightly engage the cartridge terminals.

5) Tighten the phono plug to the preamp as follows: remove the plug and squeeze the outside prongs together, so that the plug fits quite tightly; many cases of pernicious hum can be traced to a loose connection here.

6) Convert your 2-wire (inside wire and shield) input system to a 3-

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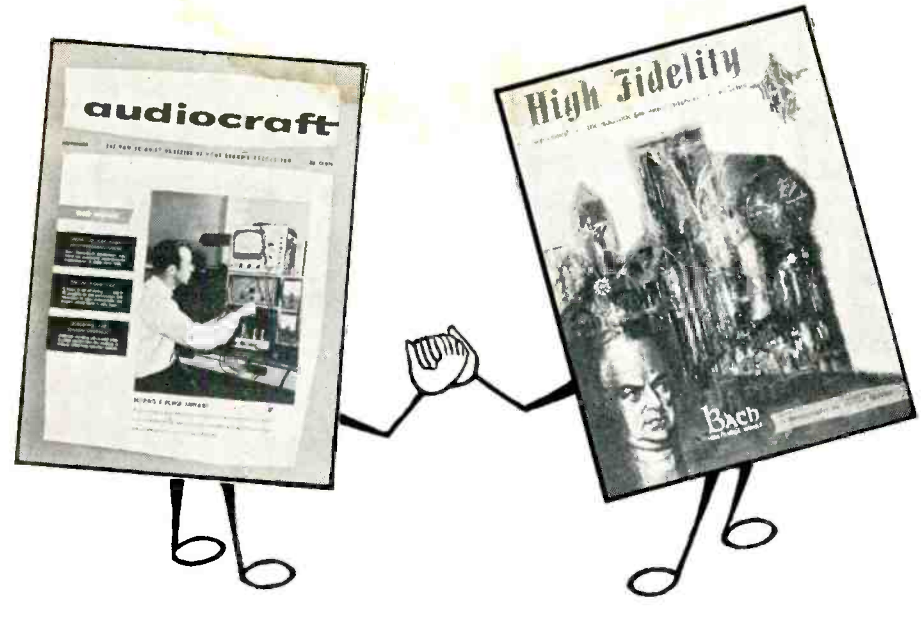

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And yes, indeed, we're "kissin' cousins" all right . . . for we're close as family members can be, yet each of us intends to maintain his own specialized individuality. *Both* of us aim to please you . . . *both* of us can contribute mightily to your enjoyment of home music in all its phases.

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Tape News and Views by J. Gordon Holt

More on Incompatibility

Last month this column painted a rather black picture of tape recorder head incompatibility, stressing the problems of incomplete erasure which arise when different head configurations are used. There are other forms of incompatibility — several, as a matter of fact: this month, let's talk about the problem of equalization. This is without doubt the most important type of incompatibility for the high fidelity enthusiast.

Everyone who buys pre-recorded tapes has a long list of reasons why he is willing to pay more for 40 minutes of music on tape than he would for an hour of the same music on disc, which he will proceed to rattle off like a catechism if slightly encouraged. But he is very often deluding himself. There is some gorgeous sound available on modern pre-recorded tapes, but the chances of hearing it well are pretty slim unless the tapes are played on a professional recorder or on one of the latest top-quality home units. Sadly, the results are often inferior to what the hi-fi enthusiast could get from records with a good pickup and equalizer. The reason is incompatibility.

As readers know, the voltage generated in a playback head by the passage of magnetized particles on the tape is a complex function of frequency. At low frequencies, voltage is proportional to frequency. As the mid-highs are approached, the voltage output of the head is affected by tape speed and width of gap. Thus the familiar graph of unequalized tape playback characteristic shows the response rising from the low frequency end of the graph at a rate of 6 db per octave. This line continues to a peak somewhere around 3,000 cycles and then rolls off. The shape of the curve is always the same, but where the mid-range peak occurs depends fundamentally on tape speed and gap width.

The first tape recorders produced in the U. S. used a playback head gap width of .0005 in., which gave a high-frequency limit of about 1,000 cycles for every inch-per-second of tape speed. For flat 7.5 ips playback response from such a "wide-gap" head, the highs had to be boosted at a rate of about 12 db per octave during the record operation, with a maximum boost of 15 db at 7,500 cycles. Then, in playback, the head required bass boosting below a turnover frequency of 800 cycles. And since just about all of these early recorders used "wide-gap" playback heads, they all required the same playback characteristic, so tapes made on one machine could be played perfectly on another as long as the heads were mutually compatible.

Then the production departments got together and figured that you could produce recorders less expensively by carrying out half of the required equalization in the record operation, and the other half in playback, since one amplifier (simplified, at that) could be used for both recording and playback. This worked out very nicely, apart from some resulting loss of signal-to-noise ratio and increase of low-frequency distortion in the final tapes. But when someone tried to play one of these tapes on a machine that did *all* its bass boosting in the playback mode — surprise! Overpowering bass that shook windows and made it possible for every incompatible tape owner to have a juke box in his living room. The man who didn't know what was going on, but believed quite understandably that tape was a better "standard" sound source than disc, blamed the ponderous bass on his speaker enclosure, his equalizer, his amplifier, and occasionally his living room. But he was still faced with the odd fact that his tapes, which sounded fine on his own machine, sounded thin on that of a friend down the street, whose own tapes sounded fine on the recorder *they* were made on, but boomed depressingly on other machines. This was the beginning of the equalization problem, which was tied up with the fact that different manufacturers used different amounts of equalization in the record and playback phases.

Then along came Ampex with its 400-series recorders and .00025-inch playback head gaps. As an isolated development, these so-called narrow-gap heads did almost as much to put home recording on the map as did surface blemishes on disc records. They cut tape costs in half, for a given level of performance, by pushing the high-frequency limit at 7.5 ips from about 7,500 up to beyond 13,000 cycles, but they also added another horn to the equalization dilemma.

A narrow-gap head normally requires — at a speed of 7.5 ips — 15 db boost at 15 kc in the recording process to play back "flat" at high frequencies and, for a signal recorded flat at low frequencies, the playback bass turnover frequency should be 1,600 cycles. Since the bass turnover needed to equalize the tape in playback (assuming the same speed) is a function only of the playback head gap width, we should expect to get reasonably flat *bass* response when we play on a properly-equalized narrow-gap machine a tape that was recorded for playback on a wide-gap machine. Unfortunately, it didn't work out that simply.

In the design of their 400-series recorders, Ampex decided that it would be useful if the same playback curve could be used for 7.5 and 15 ips speeds; this resulted in some simplification because all equalization changes with change of speed could be made in the record circuit. Now, at 15 ips a narrow-gap playback head requires bass boost beginning at a turnover frequency of 3,000 cycles, and relatively little boost of high frequencies is needed. In order to use the same playback curve for 7.5 ips, the recording equalization circuit must suppress bass and boost treble substantially. Still, the distribution of energy in speech and music is such that the tape was not overloaded at the high end with this unusual pre-emphasis; the plan was technically sound.

The only catch, of course, was that the 3,000-cycle turnover playback curve that was standard for 15 ips, and was needed to match the modified 7.5 ips tapes, was just one octave higher than was necessary to compensate for the head alone, at 7.5 ips. So tapes recorded on other machines, whether for a wide-gap or narrow-gap head, got far too much bass boost when played on an Ampex 400.

Recordists who made the seemingly justifiable assumption that a narrow-gap head with its own amplifier would produce a flat low end from tapes made for a wide-gap head were greeted by the heavy bass of denial. Those who even more justifiably believed that a tape made on one narrow-gap recorder (which used individual 7.5 and 15 ips playback curves) would play back on another, the Ampex, were equally confused at what happened when they tried it. On the other hand, the few hopeful individuals who still suspected that everything might work fine in the other direction, got thin, screechy sound when they tried to play an Ampex tape on one of the old machines. This was the result of the "tricked up" high end that was put on the Ampex tape to make it play back flat with the 3,000-cycle turnover.

If it were simply a matter of changing the playback equalization to get perfect results, there would be the obvious solution . . . a variable playback equalizer just like the ones in our present systems, only for tape. Such an equalizer would provide one position of straight 3,000-cycle-turnover bass boost, another for 800-cycle turnover, and maybe half a dozen intermediate and generally nondescript curves for really incompatible tapes.

But low-frequency equalization is only about half the story. The other half is at the high-frequency end, where head-gap widths determine the high-frequency limit. We can take an Ampex tape and play it on, say, an early Magnecorder, which uses an 800-cycle turnover at 7.5 ips to equalize its wide-gap playback head. Normally, we should expect to get a flat bottom from the narrow-gap tape, but since the Ampex one has had its highs tipped up, we shall set our equalizer for 1,600-cycle turnover, which will probably give us the nice, flat low end we are looking for. But look what the wide-gap (.0005-inch) head does to the highs. A pathetic droop, starting at about 4,500 cycles and heading downward until it takes a sharp plunge above 7,000 cycles.

So, let's see what we can do about that. One nice thing about it is that the tape hiss drops off at the same rate as the treble response, so we can figure on being able to boost the treble by a pretty large amount without increasing the noise level *from the tape*. A carefully-designed resonant network could probably be worked out to pull the highs up to a reasonable semblance of excellence, being flat out to maybe even 12 or 13 kc. Such an equalizer would boost hiss from the playback amplifier circuit, though, and might not be practical.

Then we might try working backwards. What happens when we play an old Magnecord tape on an Ampex 400, or any other Ampex built since 1952? First of all, we get boomy bass, so we change our hypothetical tape equalizer to give a 1,600-cycle turnover and get that more or less straightened out. But we have highs trouble again. To compensate for its own playback head, the Magnecorder had to push the highs up considerably at 7.5 ips, and a resonant equalizer was used to accomplish this. The equalizer produced a curve that approached 12 db per octave at the top of its peak, and then dropped off very sharply above 8,000 cycles. But the narrow-gap Ampex head needs only about 6 db per octave boost at this frequency to maintain flat response, so it plays back the Magnecord tape with a slight peak at 7,500 cycles, followed by extremely sharp cutoff above 8,000. There isn't much that can be done about it, either. The response collapse is far too sharp to enable a simple equalizer to restore it, and since the head is still reproducing tape hiss out to 10 or 12,000 cycles, any attempt to boost the missing highs would result in exaggerated tape hiss. The remaining peak could be equalized out, but smoothing it out would show up the total lack of response above there, so, under the circumstances, it is probably better to leave it in.

And that's how we stand. Tape incompatibility due to differences in head-gap width can be ironed out fairly well by a comprehensive equalizer system, but the results from an Ampex playing old

Continued on next page

TAPE NEWS

Continued from preceding page

Magnecord tapes* can never be as good, oddly enough, as those from a Magnecorder playing Ampex tapes. Without our hypothetical equalizer, present-day 7.5-ips Ampex tapes played on a Magnecorder are screechy; Magnecord tapes on an Ampex are boomy. This is not a reflection on either machine, but is simply a matter of the individual design difference between them.

I have mentioned the Ampex 400 and Magnecord PT-6 recorders in connection with this business because, first, the Ampex curve has virtually become the "standard" curve for pre-recorded tapes, since nearly all mastering is done on Ampexes, and because the Magnecorder represents an excellent example of straightforward equalization of one of the older wide-gap heads.

Meanwhile, the owner of a recorder with one of the wide-gap heads, or one in which part of the bass equalization takes place when recording, can get nothing more from pre-recorded tape than a rather poor substitute for high-quality sound. His control unit's tone controls may help to alleviate things to the point where commercial tapes are listenable, but he won't get anything like the sound that is really on the tapes. And he still won't be able to swap tapes with his tapespondents, without some mismatching in most cases.

Head Alignment

Another thing that is likely to make one machine's tapes sound pretty poor on another is head alignment, or the lack thereof. Ideally, both the record and playback heads of a 3-headed tape recorder should be so oriented that their head gaps are precisely at right angles to the direction of tape travel. All textbooks on magnetic recording make a strong point about the high-frequency losses that occur when the playback head deviates from the 90° position, and insist that maximum response cannot be obtained unless it is precisely in alignment. However, what much of the literature fails to point out is that a combination record and playback head can be far out of alignment and still produce full frequency response *from its own tapes*. Since the same gap that records the tape also plays it back, the "playback" head is in alignment with respect to the "record" head, so the mis-aligned recording plays back perfectly on the mis-aligned head. The result, to a recorder owner who runs a frequency check

Continued on page 48

*The first Magnecorders, of the PT6 series, used the then-standard .0005-inch playback head. Later Magnecorders use narrow-gap heads, with a record equalization that requires 1,600-cycle playback bass turnover at 7.5 ips, and 3,000 cycle turnover at 15 ips.

EDITORIAL

Audio Fair Report

The New York Audio Fair is the oldest, biggest, and best-attended of all regional audio shows, which information will probably surprise no one. *We're* a little surprised, though, that we managed to find enough strength to put this issue together after spending 4 days at the show. If you think a visit to an audio show is wearing, you should try exhibiting! To top it off, we had to spend an extra day in New York because we couldn't get out — all highways and railroads were rendered useless by floods.

Still, we wouldn't miss the shows for anything. There are new visitors, new exhibitors, and new products at each one. The past year was one of significant progress in audio techniques, for the number of interesting new components and improved models was even larger than usual at the recent show. A partial list is given below.

One note before we get into that: we'd like to express our thanks for the instantaneous acceptance AUDIOCRAFT has received from just about everyone who has seen it or read about it. Its enthusiastic reception at the fall shows, for instance, was quite unprecedented and, to us, wonderfully encouraging. We'll do everything we can to make AUDIOCRAFT meet your expectations.

Tuners, Amplifiers, Preamps

Audio Artisans, Inc. — Model MP 100, a greatly simplified miniature preamplifier-equalizer-control unit, available with or without power supply; the size of a miniature camera.

Audio Exchange — A custom-built amplifier, Model AE-60, using 4 KT66's as output tubes, in tapped-screen output circuit. Chrome-plated chassis and ultra-deluxe appearance.

Bell Sound Systems, Inc. — Model 2254 FM tuner, latest member of Golden Bell group. Latest models in 2122, 2199, and 2200 series have stylized metal cage covers.

David Bogen Company, Inc. — Complete line in process of integrated redesign, using matching stylized cage-type decor. Shown were Models R660 FM-AM tuner, FM50 FM tuner, DB130 control amplifier, R710 FM-AM tuner-preamplifier, and RR550 FM-AM tuner-preamplifier-amplifier.

Brociner Electronics Corp. — New Mark 10 control amplifier, Mark 30C control preamplifier, and Mark 30A power amplifier.

Conrac, Inc. — Fleetwood series 800 remote-control TV chassis, series 810 and 610B non-remote TV chassis. These have outputs for feed to hi-fi control unit as well as audio power stages. Feature is a picture definition control.

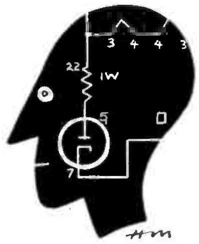
Dyna Company — Mark II 50-watt power amplifier kit designed by David Hafler has singularly impressive specifications. Price is low, appearance good.

Electro-Voice, Inc. — Five power amplifiers using new Circlotron circuit: A15, A20, A30, A50, and A100; numbers indicate power output. First two are combined with control preamplifiers in Models A15CL and A20CL; latter has new Presence control. Models PC1 and PC2 are self-powered control preamplifiers; former has Presence control. Models 3303 and 3304 are tuners with individual FM and AM sections, so that stereophonic broadcasts can be received. A control preamplifier section with Presence control is included in Model 3303.

Fairchild Recording Equipment Co. — Model 275 power amplifier, an impressive-looking 65-watter with a long list of special features.

Fisher Radio Corp. — Two new FM-AM tuners, Models 80-R and 80-T; the latter has complete preamplifier-control included. New Master Audio Control, 80-C, has tape playback equalizer, individual 5-channel mixer controls. Model 80-AZ is a 30-watt power amplifier with Z-Matic and peak power indicator.

Precision Electronics, Inc. — New Grommes GRT-1 FM-AM basic tuner; 55C, 56PG, and LJ4 control amplifiers; 211 deluxe control preamplifier; 220BA and 230BA power amplifiers, 20 and 60 watts respectively. A redesigned and expanded series of the well-known Grommes line.



Heath Company — New items include the W-5M 25-watt power amplifier kit with tweeter-saver circuit and dynamic balance control; Model FM-3 tuner kit designed to match WA-P2 preamplifier.

Interelectronics — Coronation 85 consolette control preamplifier and 40-watt Coronation 400 power amplifier, both handsome units; Constellation 100-watt power amplifier, just released.

Marantz Company — 40-watt deluxe power amplifier, completely encaged. Unusual feature is built-in meter for balancing statically and dynamically.

McIntosh Laboratory, Inc. — New MC-60 power amplifier with phenomenally low distortion specifications up to 60-watt power level.

Newcomb Audio Products Co. — Entire extensive line of tuners and amplifiers has been redesigned in an impressive way.

Pilot Radio Corp. — Newest additions are the AA-905 combined control preamplifier and 35-watt amplifier; and the HF-56, an FM-AM tuner, control preamplifier, and 35-watt amplifier on one chassis.

Pye Ltd. — Redesigned for the American market, the Pye HF-25 amplifier and HF-25A control preamplifier combination now has standard pin-tip input jacks and far more versatile equalization facilities.

H. H. Scott, Inc. — The Scott 99 amplifier now has 22 watts output, adjustable rumble and scratch filters, 2 magnetic phono inputs, and provision for tape playback equalization. Similar improvements have been made in the 210 amplifier and the 121 control preamplifier. New products this year include the 311 tuner, equivalent to the 330 combined with a comprehensive control preamplifier.

Sherwood Electronic Laboratories, Inc. — Two matching units; the S-2000 FM-AM tuner and the S-1000 Music Center combined control preamplifier and amplifier.

Sonotone Corp. — The CU-50 control preamplifier and the HFA-100, which is identical except that a 12-watt power amplifier section is added. One of the interesting features of both is that they are designed only for ceramic phono cartridges; no magnetic phono input is furnished.

Arms, Cartridges, Turntables

Bard Record Company, Inc. — The Ortho-Sonic V/4 arm is a radial arm that appears to be a practical production design. If there are no troublesome resonances — which we were unable to determine simply by examining it — the pickup tracking problem may finally be solved by the V/4.

David Bogen Company, Inc. — Two new Lenco automatic turntables: the B20 handles up to 12-inch records, has 16, 33 1/3, 45, and 78 rpm speeds, each of which can be varied within ±5%. The B50 series plays up to 16-inch records, has 16 rpm speed and is continuously adjustable from 29 to 86 rpm, and has mechanical setdown.

British Industries Corp. — The Garrard transcription turntable, Model 301, is now in full production: it features complete spring suspension and slightly variable speed around the 3 standard speeds.

Rockbar Corp. — There is a new Collaro transcription turntable, Model 2010, which incorporates an arm and a crystal turnover pickup.

Components Corp. — The Components Professional turntable is now available as the Model 70, a broadcast-type console assembly. Also available are the Pro-ette and Chairside cabinets for housing the turntable and auxiliary equipment.

Electro-Sonic Laboratories, Inc. — The C-1 (Concert) and Professional series magnetic cartridges and transformers.

General Electric — New turnover cartridges with individually-replaceable styli.

Gray Research and Development Company, Inc. — HF-400 and HF-500 (with hysteresis motor) turntables, which have steel motor boards and record cue lights, and new 108C viscous-damped tone arms.

International Audio Group — Deluxe turntable (Nikor TTO) imported from Japan has hysteresis motor and impressive specifications.

McIntosh Laboratory, Inc. — Microlab (Canada) turntables, which caused favorable comment at recent Canadian audio shows, are being distributed here by McIntosh. They feature "Floating Drive" system of decoupling between motor, base, and table.

Pickering and Company, Inc. — The new Fluxvalve turnover pickups, which

Continued on page 47

TV Can Sound Better

by John J. Huff

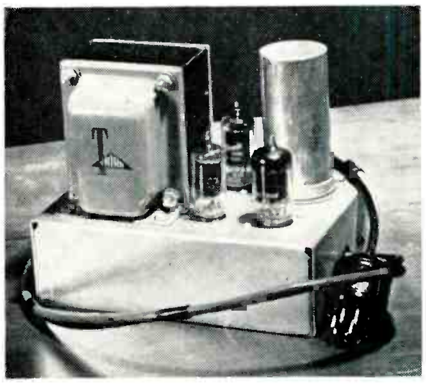
THE poor sound quality of most television receivers has been a point of concern to high fidelity enthusiasts for many years. The amplifier described here is a simple and inexpensive solution to this problem.

To keep the cost of the amplifier low and the installation simple, the amplifier replaces the output stage of the television receiver. All connections to the television receiver and power to operate the amplifier can be obtained by plugging an octal plug adapter into the output stage tube socket. This limits the maximum power output of the amplifier, because of the limited current that can be drawn from the television receiver without requiring internal changes in the set.

The amplifier develops 4 watts output or more from 50 to 20,000 cycles, and better than 3.5 watts down to 20 cycles, while maintaining a response within .5 db over this range. The current drain is only 37 ma. at 250 volts, which is easily within the power capabilities of all television receivers. Distortion is quite low at normal listening levels, comparing favorably with that of many far more expensive amplifiers. All parts can be obtained for less than $20.

Four to 5 watts seems rather low for high quality reproduction, but when it is remembered that television program material is usually of limited dynamic range, and that 4 watts is only 4½ db below a standard 12 watts, it is adequate for normal living room volume. Tests at normal volume showed that the peak power seldom exceeded one watt. Listening tests proved gratifying, for the amplifier adds much depth and clarity to the sound, and considerably improves the enjoyment of television viewing.

Normal feedback amplifier design is followed, using only simple and proven circuits. The schematic diagram is given



Thi provides hi-fi sound for standard TV

in Fig. 1. The output transformer should be mentioned because it is designed to operate with an 8,000-ohm plate load into 4, 8, or 16-ohm speaker loads. It is used here for a 16,000-ohm plate load into 8, 16, or 32-ohm speaker loads. This is necessary because good output transformers with 16,000-ohm primaries are not readily available. Transformer operation such as this is acceptable if the transformer has sufficient primary inductance to maintain the required low frequency response. The transformer listed meets the necessary requirements.

Resistor R1 is used to isolate the feedback loop from any reactive component caused by the capacitance of the shielded input cable. Failure to use this resistor could cause supersonic oscillations. Resistor R2, the feedback resistor, is adjusted for approximately 15 db negative feedback, which reduces the distortion to a very low level and leaves a comfortable margin of amplifier stability. Condenser C1 is adjusted for minimum rise time with minimum overshoot of a 10,000-cycle square wave. This will guarantee the smoothest high-frequency response. The amplifier is completely stable and free of oscillation with any value of resistive or reactive load, and recovery from overload pulses is excellent.

Actual construction of the amplifier is simple and straightforward, and should present no problems even to the novice. Fig. 2 shows the physical layout of the main parts on a 4 by 6-inch chassis, and a pictorial wiring diagram is given in Fig. 3. Fig. 4 shows a typical output stage used in many television receivers, and a diagram of the wiring of an octal plug that may be plugged into such an output stage's tube socket to make all necessary connections between the amplifier and the television reciever.

The color code of the output transformer wiring as shown on the amplifier diagrams must be followed so that the feedback will be of correct phase. If the amplifier should oscillate when completed, reverse the transformer plate connections to correct the feedback phase.

Because of the extended low frequency response of this amplifier, as compared

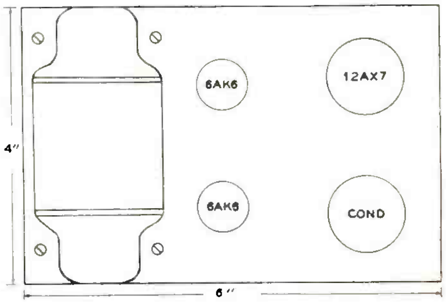


Fig. 2. Layout of parts on top of chassis.

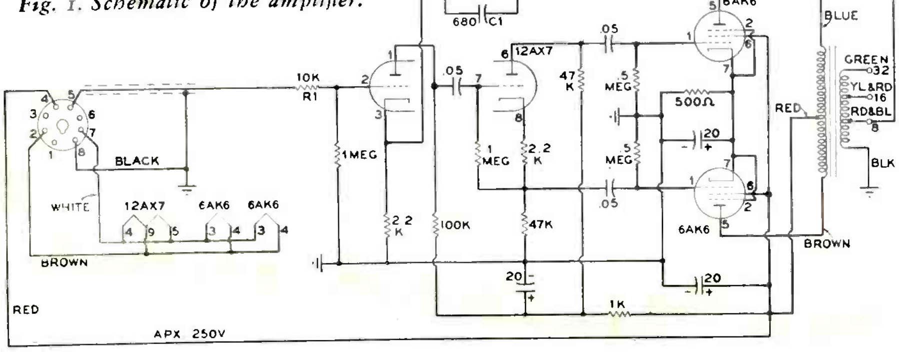
to that of most television receivers, it may be necessary to increase the B+ filtering of the TV audio stages, to eliminate hum or buzz that was not noticeable with the set's output stage. It may also be desirable to remove any condensers that shunt plate or grid resistors in the audio stages of the television receiver to ensure full high-frequency response. Do not, however, remove the normal de-emphasis condenser usually found in the discriminator load circuit. If the television receiver's output stage plate and screen voltage is much greater than 250 volts, it would be wise to lower it to approximately 250 volts by inserting a dropping resistor in the amplifier's B+ lead. The value of this resistor can be calculated by the Ohm's law formula

$$R = E/I; \quad \text{Resistance} = \frac{\text{Voltage drop}}{.037}$$

It should be of at least 2-watt size.

Since the converted output transformer's lowest available speaker impedance is 8 ohms, the amplifier should

Fig. 1. Schematic of the amplifier.



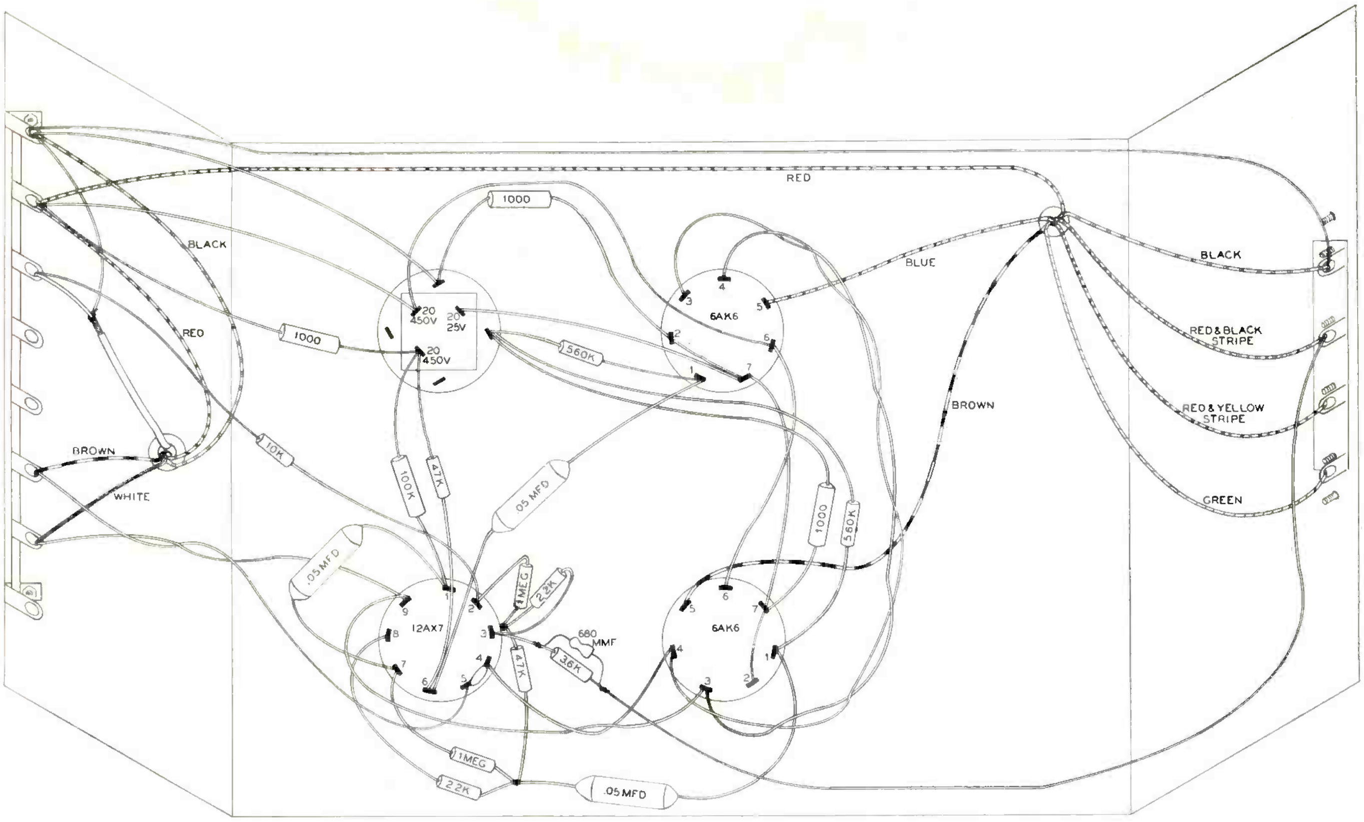


Fig. 3. Pictorial representation of under-chassis components and wiring procedure.

not be used with a speaker having less than a 6-ohm voice coil. Many television receivers have a 3.2-ohm speaker. If such is the case, it must be replaced with an 8-ohm speaker. And in almost all cases, replacement of the set's speaker with an inexpensive hi-fi speaker system will result in greatly improved sound.

PARTS LIST

RESISTORS

1	10,000 ohms, ½ watt
2	2,200 ohms, ½ watt
1	3,600 ohms, ½ watt
2	47,000 ohms, ½ watt
2	560,000 ohms, ½ watt
2	1 megohm, ½ watt
1	100,000 ohms, 1 watt
3	1,000 ohms, 1 watt

CAPACITORS

3	.05 mfd, 600 volts, paper
1	680 mmf, mica or ceramic
1	triple-section electrolytic; 20, 20, 20 mfd. 450, 450, 25 volts.

TRANSFORMER

1 Triad S-31 A output transformer or equivalent.

MISCELLANEOUS

2	7-pin miniature tube sockets
1	9-pin miniature tube socket
1	4 x 6 x 2-in. aluminum chassis
1	octal plug
1	4-lug screw terminal strip
1	8-lug solder terminal strip
	Nuts, bolts, wire, cable as required.

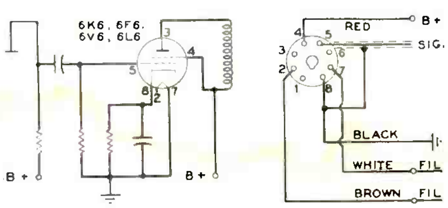
TUBES

1	12AX7
2	6AK6

AUDIOCRAFT Test Results

We found that this entire amplifier could be built for slightly less than $18 at standard net prices, and involved 4 hours' working time for an experienced constructor. It would probably take a

Fig. 4. Wiring of power and audio plug.



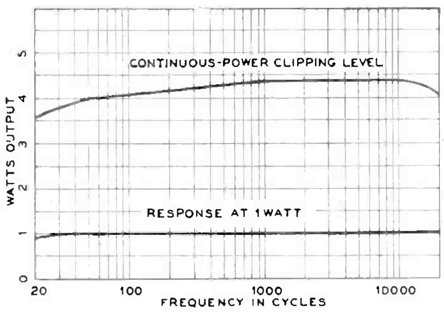
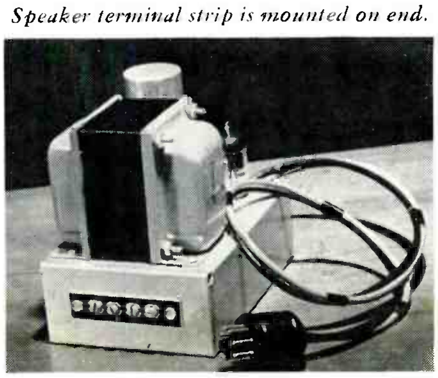


Fig. 5. Response and power performance.

maximum of 8 hours for a novice, including chassis drilling and punching.

Performance of the completed unit is so good as to be surprising. Response and power curves are given in Fig. 5; response from 20 to 20,000 cycles is almost perfectly flat. The power curve is above 4 watts over most of the range, dropping to 3.5 watts at 20 cycles. Such flatness in a maximum-power curve is quite unusual; few standard amplifiers — even expensive ones — can do this well.

Speaker terminal strip is mounted on end.

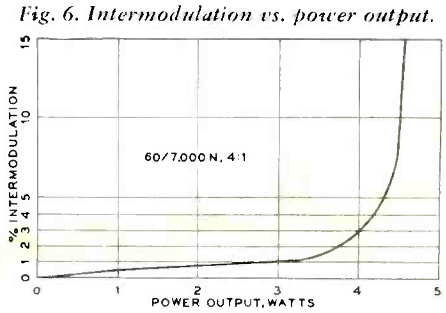


Intermodulation distortion is shown in Fig. 6 as a function of output power. Frequencies of 60 and 7,000 cycles, mixed in a 4-to-1 ratio, were used for this test. It is a particularly tough test, and the amplifier showed up very well in our opinion. At normal listening-level peaks (1 watt) the IM is below 0.5%, and at 150 milliwatts (average power used for normal TV listening) the IM is less than 0.1%! It doesn't reach 1% until the 3-watt point.

Mr. Huff has designed carefully, as Fig. 7 shows. Some ringing is evident on square-wave tests with resistive loads: the ringing frequency is in the 100,000-cycle range. When the load is reactive, however — when a speaker is connected — the ringing virtually disappears and might be considered negligible. It isn't noticeable on listening tests, in any event. The amplifier is rock-stable, with no tendency toward harsh highs or mushy bass. Damping factor is 6.85,

Continued on page 43

Fig. 6. Intermodulation vs. power output.



Speaker Enclosures: TYPES AND DESIGNS

by EDGAR M. VILLCHUR

THE best loudspeaker made is a failure, at least from the point of view of the bass spectrum, unless it is correctly mounted. Improper mounting of the speaker is probably the most common single source of trouble when the performance of an assembled reproducing system doesn't live up to the quality of its individual components.

An unmounted speaker is like a "feathered" airplane propeller. Force and motion may be present in ample quantity, but the propeller hardly engages the medium against which it beats, and allows the air to slip freely past its blades. Similarly, air pushed forward by an unmounted speaker slips around the edges of the cone, filling in the partial vacuum at the rear. The speaker makes only a small local disturbance, instead of working against the air of the room. Since this effect is most prominent at low frequencies the unmounted loudspeaker has very poor bass response, even though the cone may be observed to vibrate vigorously on low tones.

The major part of enclosure theory is devoted to just this problem. Actually, the basic theoretical solution is quite simple; all that is necessary is to cut a hole in a very large rigid board, and to mount the loudspeaker frame against the board in such a way that the speaker faces its audience through the opening. The board then acts as a "baffle" or separator between the front and back of the speaker cone, preventing the interflow of air currents that saps bass performance.

Practical difficulties arise, though, because for perfect separation down to extremely low frequencies our board must

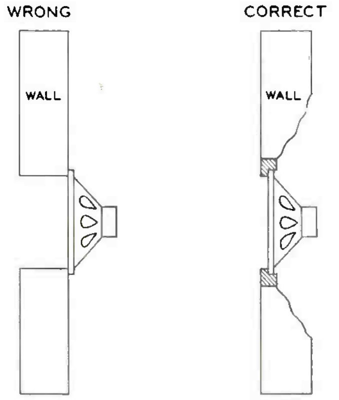


Fig. 1. Wall-mounted speaker.

be about 15 ft. on each of 4 sides. So various types of speaker enclosures have been devised, designed to keep the speaker cone from losing its acoustical bite at low frequencies without requiring baffles of unusable size. There would seem to be literally dozens of kinds of enclosures, judging from sound equipment catalogues. But all current loudspeaker mounting devices can be divided into 3 basic groups: the horn, the resonant enclosure, and the direct-radiator enclosure or infinite baffle.

Wall Mounting

The last of these is the simplest and is thought by many to be the best, but is often impractical. The most direct method of construction for an infinite baffle (by which is meant, literally, an acoustical divider of such dimensions that there is no air leakage at all around the edges) is simply to saw a hole in the wall. This provides about as excellent a mounting device as can be purchased at any price. Certain precautionary rules must be observed but, aside from the possibility of arguments with the landlord, wall mounting involves the least number of pitfalls for the amateur who is trying to achieve top-quality reproduction.

Rule 1 is that the speaker must be well anchored, either to architectural members or to a strong board of ¾-inch ply.

Rule 2 is that the speaker must not work into a long, pipe-shaped tunnel formed by the thickness of the wall. Such a tunnel acts like a short organ pipe, adding its own resonant voice in the upper bass to the reproduced music. The difficulty can be avoided by mounting the speaker flush with the front surface of the wall, and cutting away a section of wall behind the speaker, as shown in Fig. 1.

Rule 3 is that the speaker must be backed by a very large volume of air, such as would be the case if the mount-

Fig. 3, left: Infinite-baffle mounting method using an adjoining closet. Fig. 2, right: Alternative method utilizing a stair well.

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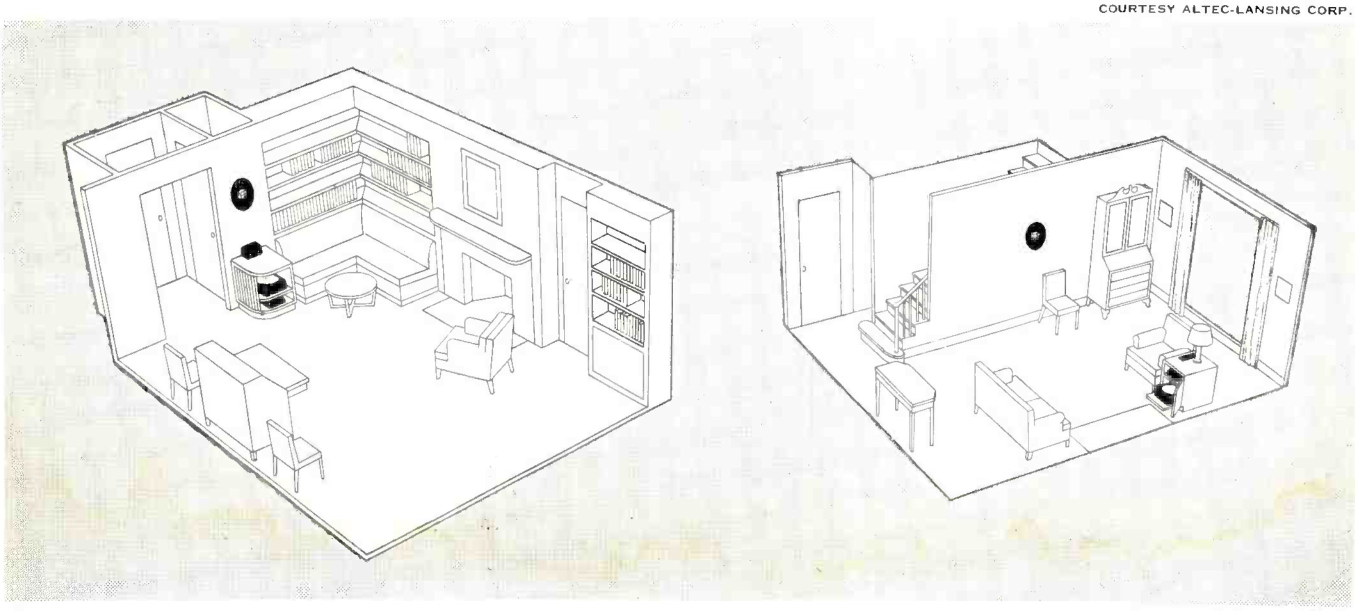
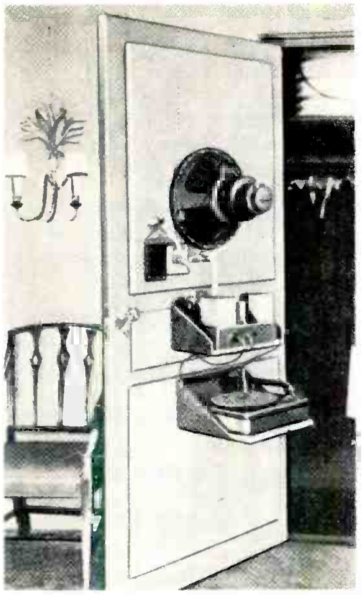




COURTESY ALTEC-LANSING CORP.

Fig. 4. Speaker mounted on closet door.

ing wall separated 2 rooms, or covered a stairwell or large closet. If the volume of air behind the speaker is limited, the principles governing total-enclosure cabinets, which are discussed a little further on, apply.

Figs. 2, 3 and 4 illustrate various arrangements for infinite baffle mounting. The room pictured in Fig. 2, it may be noted, offers an opportunity for a real mistake: if the speaker had been mounted in the wall close to the stairway entrance there would be virtually no baffling effect.

Figs. 3 and 4 show 2 methods of using a closet to advantage. When the speaker is mounted on a closet door, the thin door panels must be replaced by substantial baffle boards, and precautions must be taken against the latch rattling. Clothes on the rack not only do no harm but are desirable, because they absorb radiation from the back of the speaker cone.

Total-Enclosure Cabinets

Wall mounting is very often out of the question, for one reason or another. Infinite baffling can still be used, but ordinarily requires a very large and heavy speaker cabinet.

If we take a fairly small baffle and bend it around the back of the speaker to form a closed cabinet, we effectively prevent all interaction between the front and back radiation. One might think that this provides a cheap, foolpoof method of getting infinite baffle performance, but such is not the case. A new problem is created by the walled-up body of air, which constitutes a pneumatic springy cushion pressing against the back of the cone. There is nothing wrong with applying elastic tension to the speaker's moving system — as a matter of fact, the loudspeaker can't get along without it — but the tension has already been provided in the design of the speaker itself. The elastic air cushion is worse than a fifth wheel, because it works actively to reduce the bass range of the speaker.

The effect of the air cushion can also be explained in terms of the resonant frequency of the speaker's moving system. All speakers, whether we like it or not, have a natural mode of vibration at some frequency, like a weight on a rubber band. The lower this natural or resonant frequency the better, because below resonance progressive loss in output can be expected. Now, the greater the elastic tension on the moving system as a whole, the higher will be the resonant frequency. A speaker with an original resonant frequency below 50 cycles may have its resonance raised to 80 cycles, with an attendant loss of bass response, if it is mounted in a small, closed cabinet.

If the cabinet is large enough, however, the pneumatic spring of the enclosed air is so weak as to be almost

COURTESY R. T. BOZAK CO.

Fig. 5. Commercial infinite baffle (Bozak)

negligible. The resonant frequency of the speaker is affected very little, and the system behaves like an infinite baffle.

Such a cabinet must be extremely strong and vibration-resistant. Nothing thinner than 3/4-inch stock should be used anywhere; joints should be both screwed and glued, and all panels should be generously ribbed. Ribs made from scrap ply (cross-section 3/4in. by 1 1/2 in.), mounted edgewise and spaced every foot or so, will furnish adequate rigidity*. They are, as a matter of fact, preferable to 2 by 4's spaced every 2 ft. The problem is to achieve maximum rigidity rather than overall strength.

Inside walls should be lined with sound-absorbent material, such as Kimsul or Fiberglas, in 4-inch thicknesses. This is to prevent air column resonances from forming within the cabinet. The lining is usually tacked rather than glued on, since air spaces between it and the walls are all to the good. An alternative method is to fill the cabinet completely with sound-absorbent material. In any case the speaker itself should be protected against bits of the material, especially Fiberglas, from getting wedged into places where they would cause trouble. Such protection can be furnished by layers of cheesecloth or other suitable material placed behind the speaker frame.

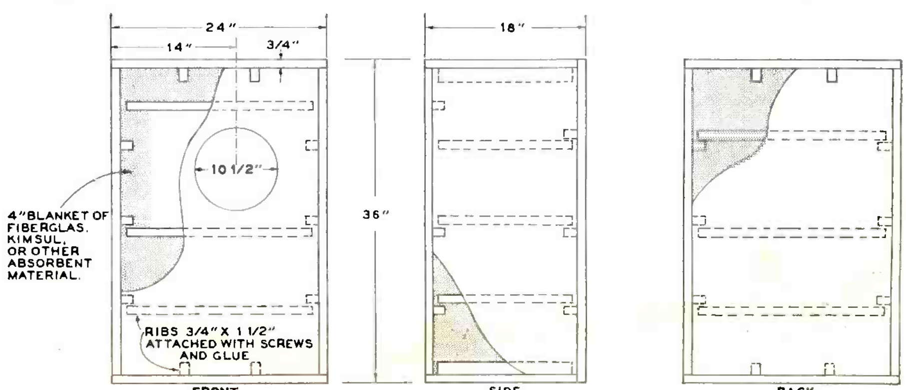
Minimum cubic volume of the enclosure required by a particular speaker depends upon various factors, which include the size and mass of the cone and the resonant frequency of the speaker mechanism. Values for typical high-quality units, however, are fairly constant. In general a 15-inch speaker should be provided with an infinite baffle cabinet of at least 15 cubic ft., a 12-inch speaker with a cabinet of 9 cubic ft., and an 8-inch speaker with a cabinet of 4 1/2 cubic ft. These are conservative figures; reduction by 25% won't result in many cycles lost in the bass. The shape of the enclosure is not important except that a long, pipe-like structure should be avoided.

One acceptable cabinet for a 12-inch speaker is illustrated in Fig. 6. The speaker opening is cut slightly off center to avoid diffraction effects that occur when sound waves reach the outside corners of the cabinet at the same time.

When the precautions outlined above are observed, the total-enclosure cabinet, like the wall installation, requires no adjustments and delivers top performance. The speaker is allowed to produce clean bass within its capabilities; boomy reproduction caused by mis-tuning, troublesome in resonant-type enclosures when the speaker and cabinet are not carefully matched, does not occur.

*For more information on cabinet rigidity see "Load It With Sand", by E. B. Schoedsack, in this issue. —ED.

Fig. 6. Total-enclosure cabinet that is suitable for any good 12-inch loudspeaker.



The acoustic suspension system (which made its appearance only last year) is an extension of the infinite baffle principle. The speaker mechanism and its enclosure are designed as a single device, one being incomplete without the other.

In this system the pneumatic air cushion created by the enclosed volume of air, instead of being a bothersome fifth wheel, is used as the main source of elastic restoring force required by the speaker mechanism. The speaker itself has far less spring tension built into it than it needs, and the pneumatic spring of the enclosure's air becomes a necessary element in the system rather than an unavoidable evil.

In terms of resonant frequencies, the speaker mechanism starts out with subsonic resonance, between 10 and 15 cycles; when the springy stiffness of the enclosed air is added to the system, by sealing the speaker into a small cabinet, the resonant frequency is raised to the desired value. The dimensions of the enclosure can be calculated to produce the final resonant frequency for which the speaker is designed.

The air cushion has been used in this way for 2 reasons. Most important is that a pneumatic spring is inherently higher in quality than a mechanical one, so that speaker distortion is reduced from that of an equivalent speaker mounted in a wall or large enclosed cabinet. The second reason is more obvious: the enclosure volume required to provide the necessary elastic tension for the present commercial 12-inch acoustic suspension speaker (with a final resonant frequency of 43 cycles) is less than 2 cubic ft. Despite the small size, however, the cabinet is functionally an infinite baffle, since the enclosed air applies no unwanted elastic tension to the speaker, and there is complete separation between front and back waves.

The "boffle" of H. A. Hartley does not, strictly speaking, fall into the infinite baffle class of enclosure, because the cabinet back is not sealed off. The difference is more semantic than real, however.

Hartley employs a fairly small enclosure. Relief is provided from the elastic tension of the air cushion by leaving out the back of the cabinet. But the entire enclosure is filled with a system of layers of sound-absorbent material which prevents the cabinet air from resonating as in an open backed enclosure. Very little of the acoustical energy radiated from the rear of the speaker reaches the front.

It will be made clear a little further on that an enclosed cavity of air, with an opening to the outside, constitutes an acoustical resonator of the Helmholtz type. The Hartley enclosure, in rigorous acoustical terms, would have to be described as a highly damped Helmholtz resonator. (The fact that it is called a resonator does not imply that all effects of the air resonance are not damped out). Functionally, however, it belongs in the direct-radiator group.

The Resonant Enclosure

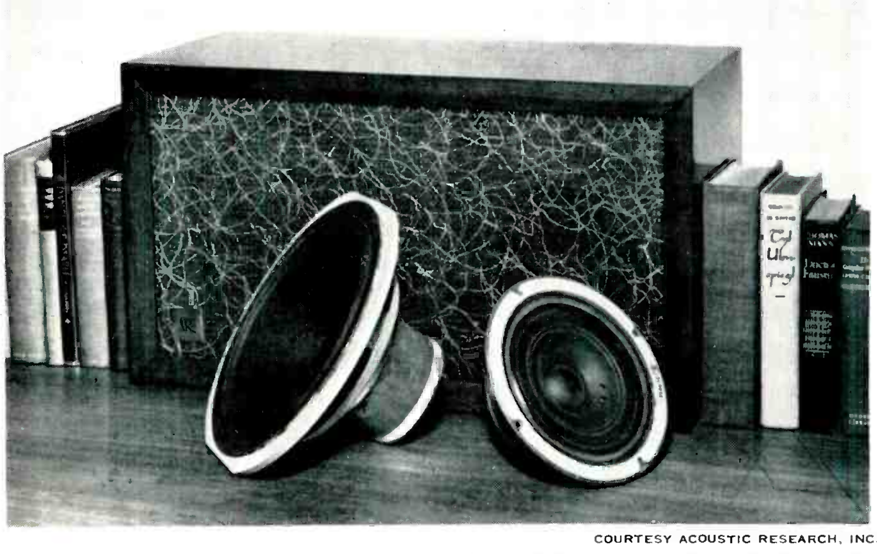
A resonant enclosure employs an acoustical resonator to extend or influence in some other way the bass range of the speaker. The characteristics of this resonator can be so adjusted that in conjunction with a particular speaker the final bass output is substantially uniform over a wide range.

There are 2 types of acoustical resonance: air column and Helmholtz. The flute and organ pipe are well known devices of the first type; the ocarina, or an empty bottle used as a whistle, the second. In either case stimulation of the resonator by some sort of acoustical energy makes it sound at a particular natural frequency of its own.

The resonant enclosure, whatever the type, allows the speaker to radiate from one of its cone sides into and through the acoustical resonator. For example, in the case of the bass-reflex enclosure, the rear of the cone communicates with the room through the Helmholtz resonator cabinet (in Helmholtz resonance the enclosed body of air vibrates as a single unit rather than in sections). The acoustical labyrinth is similar, except that the rear of the speaker cone sees the room through a long air column. In the case of the air coupler it is the front of the cone which stimulates the air column, and the speaker pumps air across rather than through the column, as in a transverse flute. The use of air column resonance involves increased problems, because unlike the Helmholtz resonator the column vibrates at harmonic overtones of its natural fundamental frequency.

Anyone who has ever blown across the neck of an empty bottle knows that the bottle will "speak" at its own pitch. The same thing happens when a speaker sends its energy through a resonator. This would seem to be undesirable for natural reproduction, but it is not as bad as it seems. The acoustical resonance can be adjusted to work with the resonant characteristics of the speaker itself, and can be "damped" (made less lively) so that the final output of the system is quite uniform.

Advantages that can be derived from this type of enclosure are that the speaker cone travel required for a given sound volume in the bass is reduced, thereby reducing distortion, and that the size requirements of the cabinet are reduced. The big disadvantage of the resonant enclosure is the difficulty and critical nature of its proper adjustment to a particular speaker.



COURTESY ACOUSTIC RESEARCH, INC.

Fig. 7. Acoustic-suspension speaker system. Note driver units. (Acoustic Research).

Tuning a Bass-Reflex

A typical bass-reflex cabinet is illustrated in Fig. 8. It should be made in the same way as the total-enclosure cabinet discussed previously, except for 2 features: the size can be cut in about half, and it contains an opening or port. The port is shown below the speaker, but it doesn't have to be at that particular location, nor does it have to consist of a single opening. The port area cannot be specified beforehand since it should be adjusted for the particular speaker used. In general, though, you should wind up with a port area roughly ½ to ¾ the radiating area of the speaker cone.

A standard setup for tuning a bass-reflex cabinet is shown in Fig. 9. The signal generator is swept over the frequency range of speaker resonance, normally below 100 cycles. With no port opening there will be a single frequency at which a pronounced maximum voltage will appear on the meter — this is the resonant frequency of the speaker as mounted. With a port, however, voltage

Continued on page 40

Chassis Layout and Wiring *by Glen Southworth*

If you are seriously interested in sound reproduction, sooner or later there will come a time when you find yourself with a soldering iron between thumb and forefinger, a chassis in the other hand, and a firm determination to create from a small stack of raw parts a working piece of electronic equipment. The enterprise in mind may stem from the purchase of a kit, complete with punched chassis and pictorial diagrams; it may be from a construction article on a device that sounds so interesting that it's got to be tried out; or, perhaps, you've felt the touch of creative inspiration yourself and have come up with an idea that, though not necessarily revolutionary, still isn't available through commercial channels.

With nearly any project you may decide upon, the degree of personal (and possibly economic) satisfaction that you achieve will depend largely on the final performance, appearance, and reliability of the completed device. These factors in turn rest on such considerations as selection of components, chassis layout, mounting of components, and wiring practices. The initial circuit design will usually be created by someone else, unless you have plenty of confidence in your electronic knowledge. However, the other matters must be decided by you as the constructor, and it may frequently be necessary for you to deal with all of them unless you are supplied with unusually detailed information.

Let's take a tough case — for example, one in which you're given the circuit diagram of a complete power amplifier, including nominal component values, and nothing else. This means that it's up to you to select the parts, get the proper size of chassis, make the layout, mount the components, and wire the thing up.

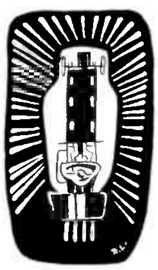
The first step is to specify and get the parts. Chances are that you're not too good a troubleshooter, and anyway you'll want as reliable performance as you can get, so the best course is to purchase high-quality items. This generally means standard brands of components, especially tubes and capacitors. Be sure that the electrical ratings of the parts are well in excess of what they'll actually be required to handle, because this will greatly reduce the chances of equipment failure after a period of use.

Unless you're pretty sure of what you're doing, it may be a good idea to select a chassis after you've obtained everything else. This gives you an opportunity to spread the various items out on a sheet of wrapping paper and decide what goes where, and how large a chassis will be required. The business of parts layout is very important, and even professionals sometimes make mistakes. The objective usually is to obtain a neat, reasonably compact piece of equipment without running into problems caused by unwanted interactions between components or circuits.

Layout

There are 3 main difficulties that can be caused by poor layout. These are heat, magnetic coupling, and electrostatic coupling. Heat usually means limited life expectancy for sensitive parts such as filter, coupling, and bypass capacitors, and, to a lesser degree, other components such as resistors and transformers. Tubes, especially rectifier and output types, are the worst heat producers, with overloaded power transformers and resistors running second. So as a rule of thumb, provide adequate space for circulation of air between tubes and other above-chassis components, and keep capacitors away from heat-producing elements.

While heat is essentially a problem relating to reliability, magnetic and electrostatic coupling can seriously affect



equipment performance from the very first moment of operation. Power transformers are usually the most serious offenders in producing magnetic fields, and can generate appreciable hum in the system when the layout is poor. Circuit elements most susceptible to hum pickup from this source are audio transformers and chokes; these should be well isolated from the power transformer and positioned for minimum hum pickup. This is especially true of input transformers or tone control inductors, because hum voltages picked up by them may be greatly amplified by succeeding stages of the equipment.

It should be noted that the strength of the magnetic field surrounding the power transformer may vary enormously, depending upon the manufacture of the unit. As a result, a layout that performs satisfactorily with one transformer may produce intolerable hum with a different type. Unfortunately, very little information is readily available regarding this characteristic of power transformers, so the policy of putting the transformer off by itself is the safest one to follow. The transformer field can actually generate small hum voltages in the chassis, as well as in nearby wiring, so it's a good idea to keep preamplifiers and input stages well away from this source of contamination.

Electrostatic fields can produce not only hum problems but serious distortion, faulty frequency response, intermittent or continuous oscillation, and pickup of electrical noises. To reduce hum pickup all AC wiring, such as the 110-volt line and the power supply high voltage winding, should be kept well away from the amplifier input stages. Shielding the input stages also helps to minimize this problem, as well as to reduce the possibility of high frequency pops, clicks, and buzzes getting into the system.

More subtle difficulties are produced when energy from the output stage of the amplifier finds its way back into the input. This can happen in 2 ways: first, if the input and output circuits are too close together, the several hundred volts of audio that may appear at the plates of the output tubes are sufficient to generate a small voltage in the input wiring, with consequent production of distortion and possibly intermittent or continuous oscillation. Usually this occurs at the higher audio frequencies, because electrical energy in this range more easily jumps the gap between input and output circuits, and with modern designs and components oscillation may take place in the supersonic range. If continuous in nature, such an oscillation will not be heard, but reproduction may be seriously distorted and limited in maximum output level. On the other hand, if the oscillation is intermittent in occurrence it may sound like speaker rattle or some other mechanical disturbance.

Of course, the cure for such a condition is to keep the input and output circuits well separated from each other and to use shielding if necessary. This means not only the tubes and associated components, but items such as volume and tone controls, switches, and input and output leads and connectors as well. Oscillations may occur at low frequencies because of feedback through an inadequately filtered power supply. But this is primarily a matter of original circuit design, and you should have no difficulty with "motorboating" if you choose a good design.

A second form of circuit interaction relates primarily to wiring techniques, and will be dealt with later on.

With the foregoing in mind, let's pick an example of amplifier layout and refer to Fig. 1A, which is supposed to represent a horrible case. The 2 can-type filter capacitors are squeezed in between the power transformer and the rectifier and output tubes, where presumably they'll be ready to serve to company after a few hours' roasting. The power transformer is nestled cozily between the output transformer, on one side, and the filter choke on the other, thus facilitating hum pickup by both these components. For good measure the 110-volt line enters the chassis next to the amplifier input jack, and the preamplifier is bracketed by the AC switch on one side and the high voltage AC to the rectifier tube on the other. To encourage assorted oscillations the input and output jacks are side by side, and leads from the plates of the output tubes are run the length of the chassis to the ouput transformer, which is conveniently located next to the preamp. As a bonus, the tone and volume controls are positioned next to the output tubes.

Now this amplifier looks reasonably neat, and it's possible that it might work perfectly if you ever decided to put it together that way — but it isn't likely to. Consider now Fig. 1B. This is supposed to represent wholesome, clean-cut, red-blooded layout. Note that the complete power supply, with all of its potentialities for hum production, is put on a separate chassis: all components are well isolated, both magnetically and thermally. If you use a long enough connecting cable you can tuck the whole power supply away in a closet or the basement. Glancing at the amplifier chassis, you'll find that it reads from left to right. That is, the input stages are located on the left-hand side of the chassis, voltage amplifiers in the center, and the output tubes and transformer on the right. This, however, is only a personal idiosyncracy, and it's perfectly legitimate to put the amplifier together in the opposite direction if you like. The main thing is that input and output circuits in this layout are well isolated, and for good measure a metal partition is located under the chassis for electrostatic shielding. With a layout like this you can hardly go wrong, except when wiring it up.

The size of the chassis will naturally depend upon how many tubes, transformers, and other parts you have to mount on it. In your first projects, it's a good idea to choose one with more-than-adequate room for everything. For average amplifiers, power supplies, and similar pieces of equipment, a standard aluminum chassis of proper dimensions is most desirable. You'll find that these are relatively easy to work with hand tools. An example is shown in Fig. 2. Normally, sizes run from about 4 by 5 by 2 in. to 12 by 17 by 3 in., providing considerable latitude in layout.

Steel chassis, while harder to drill and punch, have somewhat greater mechanical strength and are preferable for very heavy equipments such as large power supplies. Many special types of chassis are available in aluminum. For small projects, such as separate preamplifiers or other accessory items, channel-lock or similar chassis are very handy. These are actually aluminum boxes that come apart in 2 sections, providing extra convenience in construction as well as a completely enclosed chassis. Also, they are available in a number of finishes other than plain aluminum and can make an impressive appearance. Note, too, that painted chassis don't show finger marks nearly as much as those of aluminum or steel, which may tarnish badly.

Scribing and Cutting

Unfortunately, getting all of the parts bolted neatly down to the chassis in the proper places is going to require some physical exertion, beginning with the operation of putting several dozen holes of the right size and shape in an uncut expanse of metal. You won't need a machine shop to perform the necessary

Fig. 2. Amplifier with aluminum chassis.

operations; usually a few hand tools will suffice. A drill press or electric drill is nice to have if you're going to do much construction work, but it isn't essential. You probably already have most of the tools that you'll need. These include a hand drill and bits of assorted sizes up to ¼ in., a brace and a ⅜-inch bit to fit it, a reamer, hammer, center punch, screwdriver, pliers, crescent wrench, and a few assorted socket punches for making the larger holes.

I usually start by marking the general layout in pencil on the paper wrapping that covers the chassis. This includes the locations of controls, jacks, tube sockets, can-type capacitors, holes for transformer leads, etc. Then these marks are center punched and pilot holes are made with a ⅛-inch drill. These are drilled out with a ⅜-inch drill, which is the approximate size for mounting these components, or for the insertion of the screw of the socket punch. So far, all this can be done without removing the paper wrapping of the chassis, which provides protection against scratches and finger marks. When the chassis punch is used, the paper on the bottom of the chassis may be broken in order that the lower half of the punch can be screwed on, but it is still not necessary to remove the wrapping completely. You'll save yourself considerable effort, too, by selecting transformers with upright mounting, because this customarily necessitates making only one or 2 modest holes in the chassis, as compared to one very large one via hammer and chisel or socket punch.

Once the major operations are completed on the chassis, slip it out of the wrapper and place on it the components that require additional small mounting

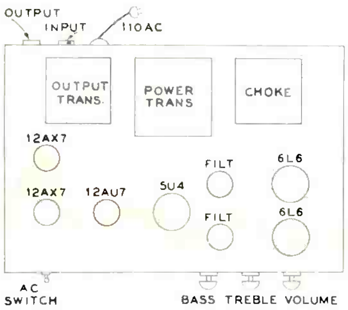
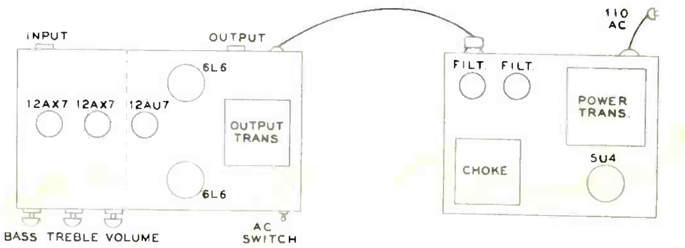


Fig. 1A, left: example of incorrect layout. Fig. 1B, right: the proper way.



holes; usually transformers and tube sockets. Locations of these holes can then be marked directly on the chassis through the mounting openings of the parts themselves. The parts are then removed, and the marks are center punched and drilled to appropriate size.

Now, a few frantic moments with nuts, bolts, and a screwdriver, and you'll be ready to lean back and catch your breath, after you plug in the soldering iron. This should be light and small enough to wield precisely at close quarters. Near at hand should be a supply of 40/60, 60/40, or 50/50 rosin core solder (depending on what you can get) and a coil of plastic-covered or push-back tinned wire, No. 18 or 20 gauge solid.

Wiring

Most experienced constructors begin by wiring the power supply. Possibly they feel that this part of the equipment should be done while you're still fresh, and less apt to make mistakes that might result in blown tubes and capacitors or burned-out transformers. Or they may simply want to dispose of the stiff, heavy wires from chokes and transformers. After the power supply is wired, tube filaments should be connected, followed by most of the point-to-point wiring that doesn't involve small parts such as coupling capacitors and resistors. They are installed last.

Tie strips, a whole handful of which may be purchased for a dollar, are very useful in mounting loose capacitors and resistors for purposes of mechanical rigidity. AC pairs (high voltage and filament leads) should be kept close to the chassis, away from other wiring, and should preferably be twisted to reduce the electrostatic field around them. The push-back wire mentioned earlier is used to connect components with no leads of their own. The wire is simply cut to length, the insulation pushed back for ¼in. or so (being careful not to run the tip into your finger) and the lead is crimped and soldered in place.

If you took my advice and built the power supply on a separate chassis, you'll need a cable to connect it to the power amplifier. This should preferably be of stranded wire, since it will stand much more usage without breaking than the solid kind. Furthermore, in order to avoid a juggling act when moving things about it's a good idea to make the cable detachable at one or both ends. I usually place a female socket on the power supply chassis for this purpose, because an exposed male plug carrying high voltage can lead to painful incidents.

The amplifier section is next, and an extra measure of caution, possibly a suspicious outlook, may be of benefit. The instructions given for the power supply apply here also, and in addition short, direct connections should be used. Resistors and capacitors should be mounted close to the associated tube sockets, and tie strips should be used whenever necessary to insure mechanical rigidity.

If you'll look carefully at nearly any circuit diagram, you'll notice that a lot of circuits seem to connect to either of 2 places — ground or B+. This is the logical time to mention the matter of ground loops, some of the trickiest and most exasperating phenomena that you're likely to encounter. They are caused by the fact that a seemingly solid and honest piece of wire is actually as much a resistor as the little tubular gadgets that you've paid hard money for. In most amplifier circuits this fact is of negligible importance, but when we get back to the input and preamplifier stages things become so sensitive that it's a good idea to be cautious. Hum is the most commonly recognized symptom of a ground loop.

You may get a ground loop by trying to skimp on the filament wiring, using ground as one side of the circuit. The center tap of the power transformer high voltage winding may give trouble, too, if fastened far away from the negative connection to the filter capacitors which, of course, should be located well away from the input stages. The magnetic field of the power transformer may induce strong hum voltages in the chassis. For this reason some constructors prefer to use a separate ground system of heavy wire, which touches the chassis at only one point for purposes of electrostatic shielding. Figs. 3A and 3B show respectively a horrible example and a separate ground system of sterling character. The input stages of a high-gain power amplifier are shown in Fig. 3A. Hum is produced because the filament supply and the cathode of the input stage are returned to ground through the same wire (indicated by X's). Another ground loop is produced by wire Y, which is common to both input and output of the amplifier: it may cause distortion or oscillation. In Fig. 3B, on the other hand, a ground system is shown arranged for minimum interaction. The ground bus may be a single piece of heavy wire connected to the chassis at one point only. All filament wiring is isolated except for the center tap of the transformer filament winding, which is grounded to reduce the AC field. Incidentally, this separate ground system business also means that it's a good idea to isolate input jacks and connectors from the chassis.

Distortion is a subject that I haven't heard mentioned in connection with ground loops, but remember that when you're trying to impress the neighbors with the latest recording you may be running an ampere or so of audio current through the ground system of your amplifier, and if some of this current gets back into the input, via injudicious wiring, some strange results may be obtained. Even if it's only a matter of 0.2% instead of 0.1%, that step from 0.2 to 0.1 isn't easy to get and there's no sense in throwing it away.

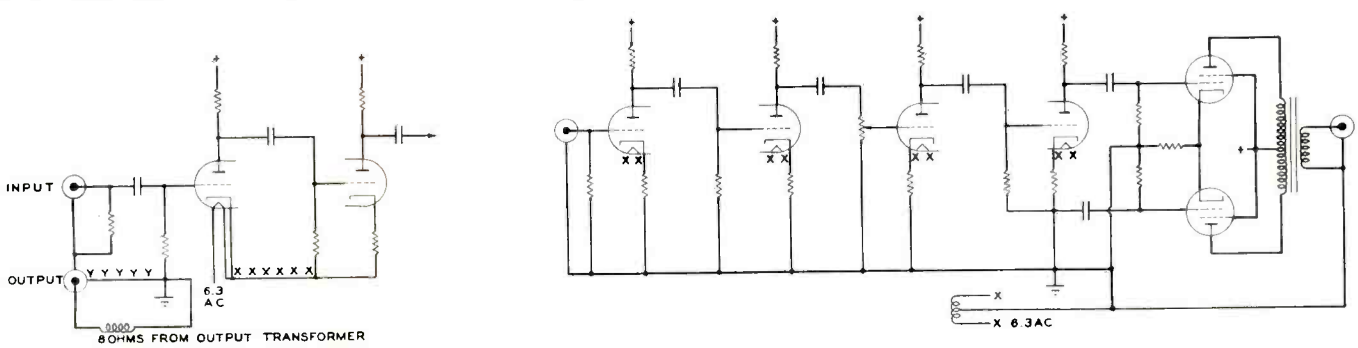


Fig. 3A, left: Wiring procedure producing ground loops and possibly objectionable hum. Fig. 3B, right: Recommended method.

After finishing the wiring, re-check everything to be sure that the diagram has been faithfully followed and connections are firmly soldered. Take a close look to see that blobs of solder haven't wedged themselves into strategic positions and caused short circuits, nor have scraps of chassis material or ends of wire. Lift the equipment into the air and shake it firmly several times to dislodge any odds and ends that you may have missed. Replace any parts that drop out during this process, then put the tubes in the right sockets, lean back, and think.

Now you've come to the point when you plug the AC cord into the wall and turn on the switch. Probably you're anxious to hear beautiful music, so you already have the speaker and turntable

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A High-Quality Microphone Mixer

by J. Gordon Holt

NOT too long ago, it was propounded by audio engineers that the human ear was incapable of detecting distortion below 2% total harmonic content in reproduced sound. This point of view was probably based on the results of listener tests conducted with pure sine waves, to which harmonics were added in measured amounts until they became audible to the listeners. As an engineering rule-of-thumb, the concept has hung on doggedly even to the present day of aural enlightenment. When we view this rating in terms of the probable IM distortion that would accompany a 2% harmonic content, however, we find the statement in question asserting that a listener is oblivious to IM distortion of 8% or less. The absurdity of this is quite evident to anyone who has traced his fuzzy record reproduction to a preamplifier whose IM has risen from 0.1 to 0.2%.

Current high fidelity equipment trends seem to reflect the viewpoint that *any* distortion is audible in the reproduced sound, and playback equipment is being designed with unmeasurable intermodulation distortion over most of its operating range. But the recording industry is still, generally, limping along on pre-war equipment standards, and using amplifiers with distortion ratings that would horrify a truly dedicated high fidelity enthusiast.

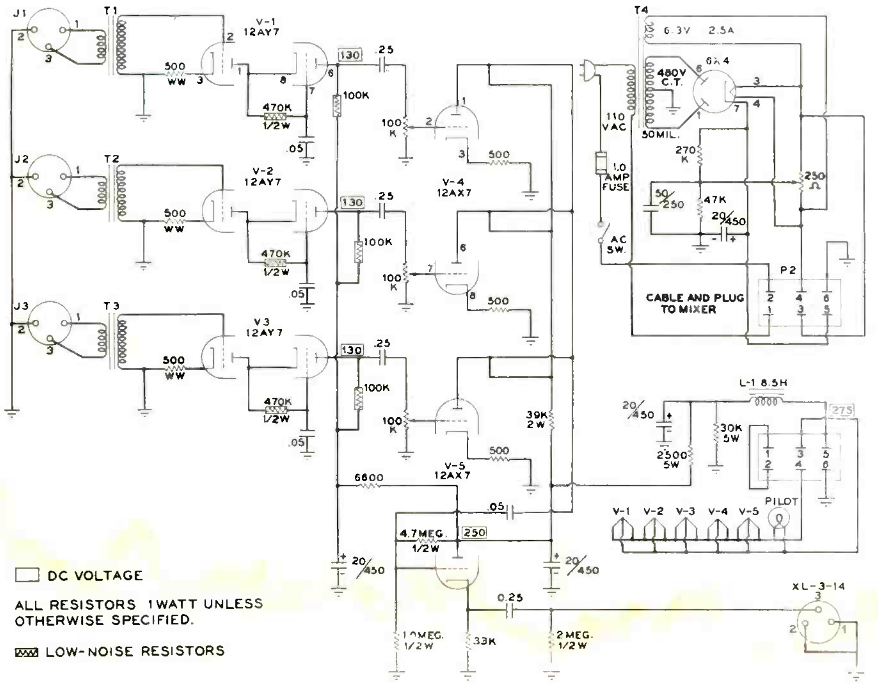
It seems reasonable that the major reason why small amounts of distortion are so annoying in record reproduction is that they aggravate the audible effects of the reproducing stylus rattling around in the groove at high recorded velocities. Such an initial source of ragged distortion does not exist to nearly as great a degree in the recording end of the chain, which is probably why record companies have not felt obliged to get the distortion in their own equipment down to the miniscule levels that are considered imperative for record reproduction. As a result, a great deal of equipment being used for professional recording purposes has far too much distortion to be considered acceptable by today's standards. Some commercial record companies are, in fact, using the same microphone mixing systems that were designed before World War II for AM broadcast use, most of which are quite blandly rated at 2% harmonic distortion.

Coincidentally, this figure happens to be the typical distortion rating for professional-quality tape recorders under *peak* recording conditions; average recording levels produce considerably lower amounts of distortion. Yet the 2% rating is apparently taken as the standard of excellence by the majority of mixing equipment manufacturers.

I have been doing a great deal of live music recording with an Ampex 401 during the past few years, and had reached that stage at which I felt single-mike pickup to be a bit inadequate when I started delving into input mixer catalogues. The revelation was not encouraging. Commercial input mixers are definitely expensive, and their quality is generally not, as far as I am concerned, concomitant with their price.

The mixer that will be described here was designed to meet the need for a truly low-distortion, wide-range portable unit that could be constructed at less cost than that of a good microphone. The following rather vague specifications were set up in advance for the projected unit: First, its intermodulation distortion at maximum operating levels had to be no higher than 1%, and at average levels as far below 0.5% as was practical without jacking up the cost too much. Second, its over-all frequency response had to be sufficiently wide and smooth that it would not in any way restrict the very wide range of the best microphones available. Third, it had to have low enough noise and high enough

Fig. 1. Complete schematic of the mixer. Two sections are connected by a Jones plug.



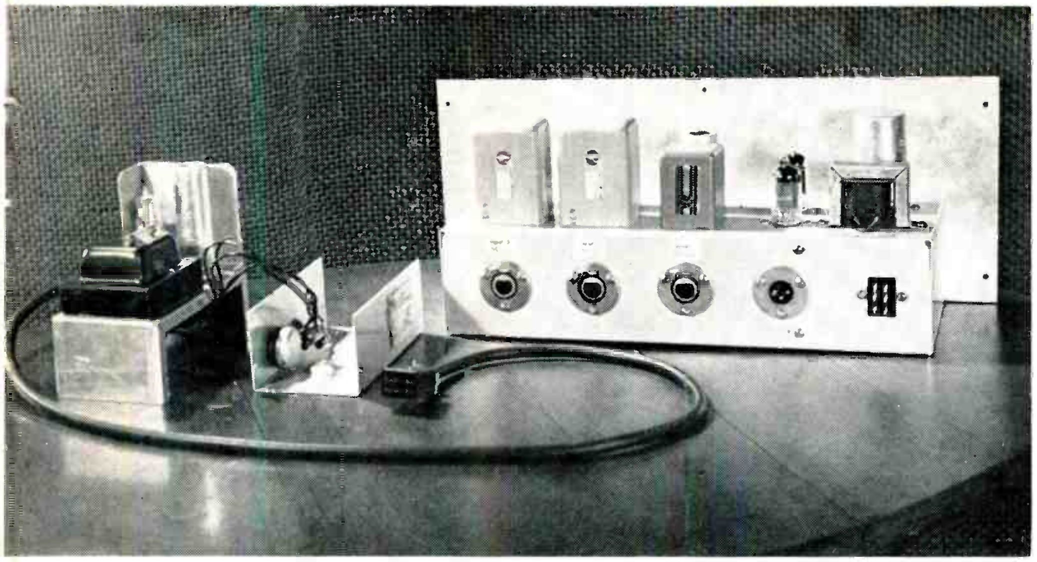


Fig. 2. Power supply with cable, left, and amplifier section with attached front panel.

it was for me), you can purchase them one at a time, using slightly less ambitious ones of similar dimensions for the remaining channels, and replacing them later if desired.

A choice of single-ended or push-pull output is offered in the secondary winding of the K-241-D transformer, as it is in many similar matching transformers. Best results are obtained with the 2 halves of the secondary winding connected in parallel, with the top of one winding strapped to the top of the other, and the same for the bottoms. See the wiring diagram enclosed with the transformer for the correct pin connections, and for the required pin strapping for the primary winding. If an input transformer is used that has a center-tapped secondary instead of a 2-section winding, use the center-tap as the ground connection, leaving the other end disconnected.

The rather large-value coupling condensers from the preamplifiers to the mixing controls are to maintain extended low frequency response into the controls, which were set at 100,000 ohms to reduce the possibility of high-frequency losses at intermediate control settings. The controls are standard audio-taper carbon units.

A major problem in the design of the mixer stages was getting the maximum possible input voltage-handling capability without fringe overload, and maintaining low distortion at lower operating levels. The following stage is a cathode follower; since there is no phase rotation in this stage, inverse feedback could not easily be applied around the mixers. So the cathode bias value

gain that it could be used with low-output microphones picking up low-intensity sounds. Fourth, it had to have sufficient input latitude to permit the use of high-output mikes under high-intensity sound conditions. And finally, the output impedance from the mixer had to be low enough to prevent cable losses at high frequencies, and high enough to permit direct connection to most existing recorders without the necessity for additional impedance-matching devices.

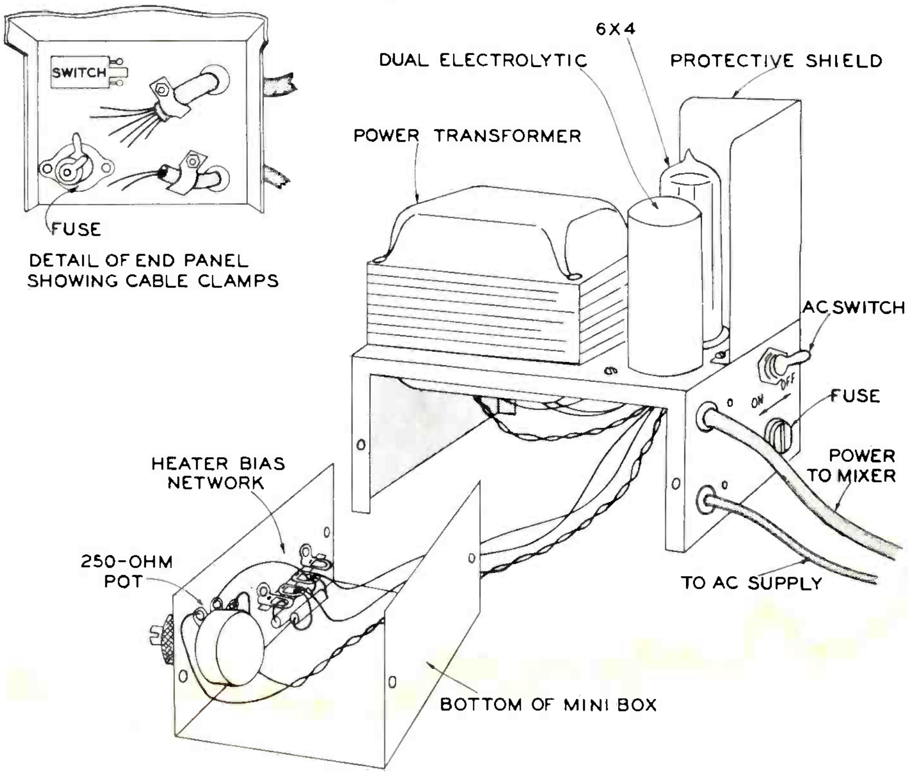
Circuit Development

Two types of input preamplifier stage were tried and discarded before the present cascode circuit, Fig. 1, was adopted. Pentode preamplifiers were, it was felt, too subject to noise and microphonics and too sensitive to hum from adjacent AC wiring. Also, their gain when operating into a fairly low-impedance grid circuit was limited. A straight 2-stage triode voltage amplifier also tended to be microphonic with unselected tubes, and it shared the pentode's additional shortcoming in being unable to handle large input voltages without overload. Inverse feedback was tried with the dual-triode stage but, by the time the voltage-handling capacity was up into the safety zone, the gain was lower than desired.

The cascode stage was finally adopted because it lacks the shortcomings of the other two systems, and also because its distortion is extremely low by virtue of the circuit's inherent degenerative action. Intermodulation distortion in the preamplifier circuits remains below 0.1% over the range of most input signals that would be fed to them, and for average signals the IM is just a little under 0.05%.

Input transformers used are Peerless 20-20-Plus units, which have performance characteristics that far exceed those of any microphones presently available. But they are very costly. Together, they are likely to account for more than half the total cost of the mixer, but I feel that their performance characteristics warrant the expense. Frequency deficiencies in microphone input equipment are most often traceable to shortcomings in the input transformers used, and there is no point in using the best microphones with associated equipment that defeats the quality the mikes are capable of. If the mikes that are to be used with this mixer initially are not of the highest quality, there is really no need for the most expensive input transformers. If the initial cost of 3 Peerless K-241-D transformers is considered too much for the budget to bear in one fell swoop (as

Fig. 3. Construction details of the power supply in mini-box or channel-lock chassis.



was carefully set for optimum performance with the existing plate supply voltage. This should be as close as possible to 250 volts, measured at the plate of the cathode follower.

The original mixer design specified a 600-ohm output transformer to feed a standard balanced line. But some investigation of recorder specifications indicated that most of the available units have a bridging input or a standard high-impedance high-level input connection, so the line transformer idea was scrapped in favor of a cathode follower, which considerably reduces the cost of the mixer. The output impedance is still low enough to allow up to 40 ft. of cable to be used without high frequency attenuation in the audible range. Distortion in the cathode follower is extremely low, and it is incapable of being overloaded by any signal that the earlier stages can feed to it.

For minimum hum, the power supply was placed on a separate chassis (Figs. 2 and 3), which was mounted in the bottom section of the carrying case. A 6-conductor cable carries the power supply voltages to the main chassis, and it is terminated by a socket at the chassis end. The plug is a flanged type set into the rear of the chassis.

The 110-volt AC pins from the 6-conductor power supply plug are strapped together in the main chassis, putting this shorting strip in series with one side of the AC line to the power transformer. The purpose of this is simply to prevent the unit's being switched on without the chassis connected, with consequent overload and popping of the power supply input filter.

It was felt also that hum could be further reduced by restricting all 110-volt AC lines to the power supply chassis, so the AC switch was located there. Subsequent tests have shown that the mixer is remarkably insensitive to hum pickup within the main chassis, probably because of the relatively low impedances at which the entire system operates all the way through, so the AC power switch could quite safely be installed on the front panel of the mixer itself — either next to the pilot light or above the chassis level, vertically over the pilot light bracket. The switch would then replace the jumper across the power supply plug, and would serve the same purpose.

Since the power transformer is physically removed from the input transformers there is no necessity for using an electrostatically shielded unit, and one of the inexpensive receiver replacement types can be used. The Merit P-3047 transformer specified in the parts list exactly meets the voltage and current requirements for this application, but any other identical transformer will serve the purpose equally well.

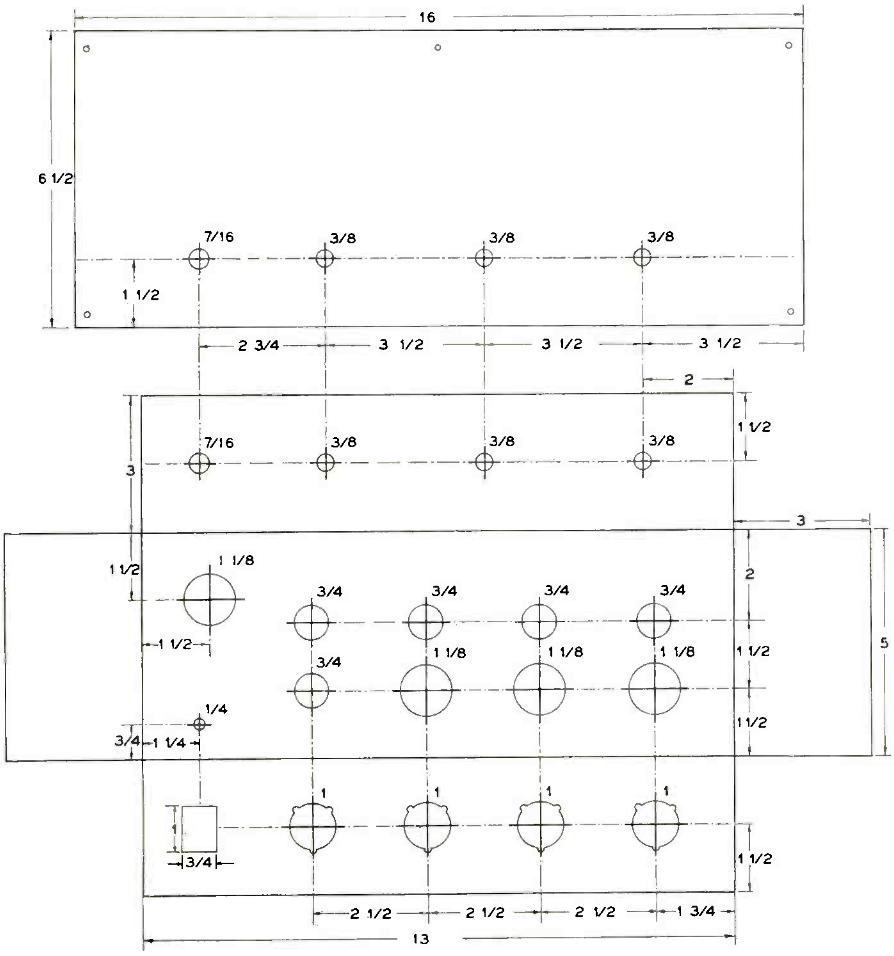


Fig. 4. Drilling plan for amplifier chassis (bottom) and attached front panel (top).

The mixer draws 2.25 amperes heater current, so voltage loss in the power supply cable can become significant unless its length is kept fairly low. Recommended interconnecting cable length is 5 ft., which is quite long enough to allow the power supply to be placed under a table on which the mixer is set up for use.

Construction, wiring

The mixer occupies an aluminum chassis 13 in. long by 5 wide and 3 high. The front panel is a 6½ by 16-inch heavy-gauge aluminum sheet, which is held to the chassis by the nuts that secure the controls and pilot light.

Fig. 4 gives the dimensional layout of the panel and mixer chassis for the components specified on the parts list. Before attaching the front panel to the chassis, any necessary surface finishing should be completed. If the panel is already finished in hammerloid or crackle enamel, all it is likely to need is filing of the cut edges and perhaps some paint retouching here and there. If it is an unfinished aluminum panel, it may be brought to a high luster with fine steel wool, by rubbing briskly along the length of the panel. Circular or cross-strokes will leave weird swirl patterns that are not overly attractive. After a good rubdown with the steel wool, the panel should be thoroughly washed with soap and water, then completely rinsed and dried. If decals are to be used for panel markings they should be applied next, and as soon as they are thoroughly dry the whole front surface should be sprayed liberally with a clear plastic protective coating. Lay the panel flat when spraying, and leave it that way until dry, so the wet plastic will not streak or run.

It is suggested that the original parts layout (Fig. 5) be followed fairly closely. While the mixer has not shown any tendency at all toward parasitic oscillation or hum pickup from stray wiring capacities, there is always the possibility that haphazard layout might lead to trouble from these sources. No shock mounting is necessary for the preamp tube sockets; the cascodes seem to be microphonically insensitive. They should, however, be shielded so as not to tempt fate from hum fields.

No shields are required on the 12AX7's, although it is suggested that a wire clamp be used over the top of the 6X4 to prevent this somewhat top-heavy tube from sagging sidewise in its socket when the mixer is carried. In Fig. 3 a small shield can be seen on the corner of the chassis, next to the rectifier tube. This is simply to protect the tube from cables and plugs stored in the bottom of the case when it is packed up.

The power supply is assembled in two sections. The channel half of the minibox chassis contains the heater bias supply and its associated potentiometer, and the cover section that fits over the channel is used to mount the power transformer, rectifier, AC switch (if not on the main chassis), electrolytic can, and cable grommets.

Leads from the hot side of the input transformer secondary to the preamplifier grids should be kept as short as possible, because they are more likely to be a source of hum pickup than any other part of the entire mixer. To accomplish this, the input transformers and preamp tubes should be oriented so that the input lead connections from the preamps (pin No. 2) to the transformers are directly adjacent. This can cut the lead length down to 1½ in.

Wiring in the mixer is point to point wherever possible, with tie-lug strips used only to anchor lead junctions that would otherwise be floating. All AC leads in the mixer chassis should be twisted and kept as close to the chassis as possible. Those running to the pilot lamp and, if it is in the main chassis, to the power switch, should be laid into the overhanging flanges around the bottom of the chassis or pushed into the right-angle fold between the top and end of the chassis. It is not necessary to shield these leads, however.

Since the rest of the wiring is largely point to point, there is no need to shield anything there either. The only signal leads likely to run to any length are those from the moving arms of the mixing controls, and as long as these leads are kept against the chassis and not closely parallel to any heater pairs, they will not pick up hum.

The aluminum chassis with its separate power supply will not be subject to any appreciable eddy current activity, so grounding of components is done directly to the chassis wherever convenient, rather than to a common bus bar. All ground points associated with the input circuits and preamplifier stages are, as a precautionary measure, carried to a single solder lug, which may conveniently be tied to one of the input transformer mounting bolts.

Some of the resistors specified on the mixer schematic are rated at ½ watt, as is the convention for grid circuits and low-current supply networks. Since the original mixer was constructed, I have changed my mind about their advisability, for ½-watt resistors seem to have a tendency to drift from their original value even when negligible amounts of current are flowing through them. For the ultimate in trouble-free, stable performance, then, it is suggested that no resistors of less than 1 watt be used anywhere in this mixer — or, for that matter, in *any* non-commercial piece of audio equipment.

The carrying case shown in Fig. 6 was made from ½-inch plywood, nailed and glued together. It was finished with a coat of grain filler, followed by two coats of dull black paint. The sections are held together by eight suitcase-type hook clasps, bolted to the edges of the sections after they were planed smooth and even. Finally, some rubber feet have been added as embellishments, with four of them on the bottom of the case (when closed and standing up), and another four on the side of the mixer section that becomes its bottom when set up for use.

Phasing

One of the most important aspects of mixer construction is maintenance of proper phase relationships all the way through the system. Microphones that are operated out of phase with each other can create the same high distortion and violent response peaks and dips that come from misphased loudspeakers.

The initial transient from a kettle drum, for instance, is a sharp rarefaction of the air that will cause the diaphragms

Fig. 6. The mixer in its portable case.

of 2 microphones to move outward, towards the source of the sound. If these microphones are correctly phased with relation to each other, they will produce a negative voltage impulse. But if the leads from one of them are reversed, that mike will put out a positive pulse of about the same magnitude as the negative pulse from the other, and they will tend to cancel, leaving nothing but spurious distortion components that were unique to each microphone and were therefore not cancelled out. Also, as the inevitable result of oddities in room acoustics, the sound pressure arriving at one microphone for a wide range of frequencies will be full of peaks and dips, however slight, that differ from the nature of the response curve "seen" by the other mike. If the mikes are in phase, the curves will tend to average out, smoothing the entire response. But if they are out of phase with each other, there will be considerable cancellation at those frequencies where both mikes produce the same output voltage, and much less cancellation of frequencies that are produced unequally from both mikes. The end result is a formidable array of peaks and dips that add to the dirtiness from the accentuated distortion.

It is fairly obvious that if switching the phasing of one microphone within the mike can cause this trouble, the reversal of one pair of leads in the mixer can do precisely the same thing. The resulting aural confusion from certain combinations of mikes and input channels can be utterly baffling and impossible to track down unless the matter of phasing is taken into consideration.

The logical place in the mixer where misphasing would be likely to occur is at the connections to the input transformers. If these are all identical units, it is a simple enough matter just to wire them identically, paying special attention to the primary circuits where crossed leads may easily go unnoticed. When using different input transformers, the problem becomes a little more involved, because pin connections vary widely from one transformer manufacturer to another. In this case the best thing to do is wire them all for the correct impedance values, temporarily ignoring the

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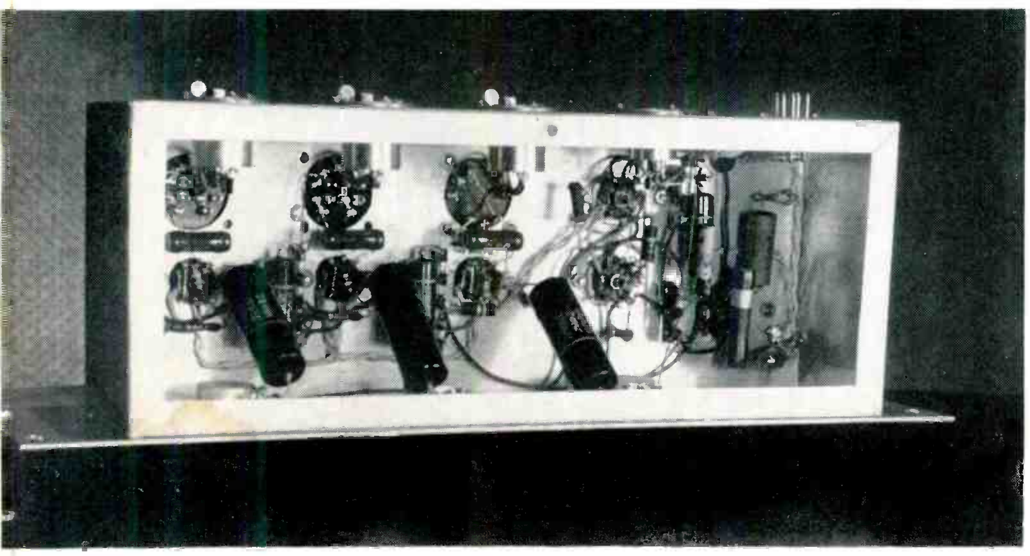
Fig. 5. Bottom view, amplifier wiring. Note placement of .25-mfd coupling capacitors.



TURNTABLE . . . *with strings attached*

by S. R. Williams

OF the components that make up a sound system capable of real fidelity, the changer and turntable seem to have been slighted by our audio engineers. Judging from the plethora of new designs for amplifiers and preamplifiers in the current magazines, they evidently prefer to concentrate their energies in the more familiar field of electronics. Yet, for playing records, the turntable or record changer remains the first and frequently the weakest link in the chain of reproduction. (Disagreement may be expected at this point from the how-inefficient-is-the-speaker contingent, but let's go on as if nothing had happened.) A close look at these devices may be expected to disclose their virtues and faults, while perhaps suggesting cures for the latter.

Although both changers and turntables are essentially machines for rotating a record under a stylus, the first has the advantage of being able to play several discs without attention from the operator; some types, resembling 19th-century clockwork mechanisms, even turn each record over in turn. In addition, the changer and automatic turntable (a cousin of the changer — operation of the arm is automatic, but it does not change records) are usually gentler with our precious records than we shaky-fingered humans in setting the arm down. However, the mechanical complexity of the machine and the necessity to keep cost down require that some sacrifice be made in workmanship, with a resultant loss in fidelity. The changer, with competition to meet and many functions to perform, is often a collection of pressed, stamped parts that can produce enough assorted noises to make it unacceptable to the perfectionist.

The turntable, on the other hand, possessing the disadvantages of requiring manual operation and being generally more expensive, is likely to add fewer extraneous noises of its own to the music.

The noises that trouble the changer and cheaper turntable are classified as *wow, flutter, drift,* and *rumble,* the last being characterized by recurrent low-pitched sounds audible during quiet passages in the music, while the others are all differences in pitch caused by speed variations in the motor or associated mechanism. These, in turn, are caused by eccentricity or slippage in pulleys and/or idlers and/or rotor and/or turntable.

In poorer designs, an interaction of the following sort is frequently encountered: the motor is out of balance and vibrates, transferring its vibrations through idler and pulley to the turntable, where it is picked up by the cartridge along with the recorded matter. At the same time, the motor vibrations have traveled through the base and up the arm to the cartridge, creating a pyramid of noise with the pickup at the apex. To make matters worse, the other parts — bearings, pulley, idlers — often add their own noises to the original.

The average beginning high-fidelity enthusiast is usually not too concerned about changer noises, so pleased is he with his equipment and the music it reproduces; but as he adds better-quality components to his system, particularly those responsive in the bass region, he will gradually become aware of the fact that something must be done about the changer if it is to keep pace with the sound quality of the rest of the system. If his is an experimental nature he may try such expedients as improving the cushioning of the changer base, rubber-mounting the arm, or installing a rumble filter. While these tricks may bring about some improvement, they will generally fall short of the ideal. The rumble filter is especially inadequate, since it inevitably removes some low-frequency components of the music as well as the rumble. It also seems pointless to strive for perfection in the bass with folded corner horns and 18-inch woofers that are guaranteed to reproduce the most subterranean organ pedal notes, only to defeat their purpose with an electronic gadget. But to have your cake and eat it too, it may not be absolutely necessary to replace your present unit with the most expensive turntable on the market. You can, in some cases, simply add weight or mass to the existing facilities.

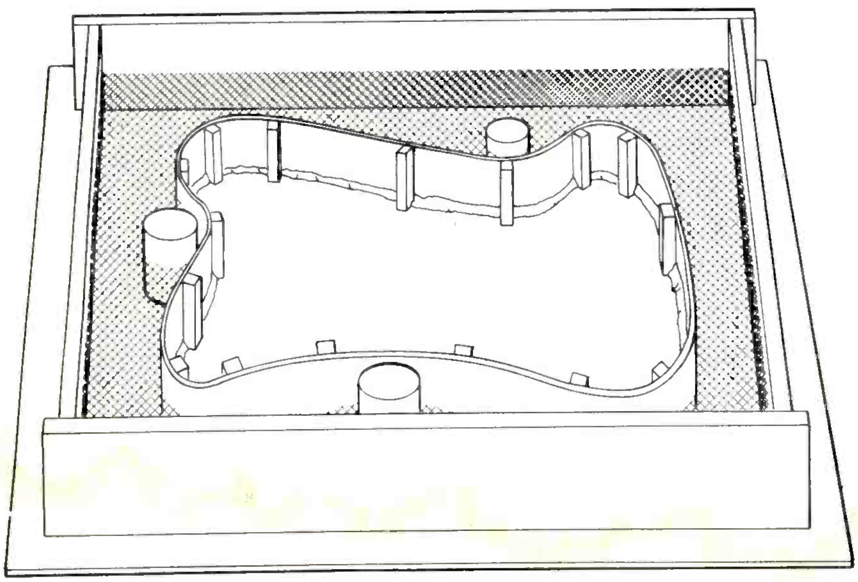


Fig. 1. How to make a poured-cement base for your present changer or turntable.

Noise Reduction by Inertia

The principal of using mass to reduce noise is used in two ways. The first is to apply the mass to the changer or turntable base plate. One writer has suggested cutting the base from a thick plate of steel and firmly bolting the machine to it without the mounting springs. While this solution would certainly result in a nice, quiet machine, because all noises would be lost in the great mass of the plate, the average man might find it a bit impractical. Cutting would have to be done by a welder or machinist in a time- and money-consuming operation; steel that thick would be as costly as an evening at Ciro's. Marble could be used, since it is relatively cheap and most types can be cut with ordinary wood-working tools. A friend of mine found the marble top of a Victorian night-stand in a junk yard, bought it for $1.00, and jig-sawed it to fit his changer which is now very quiet.

An easier solution, Fig. 1, is that

Fig. 2. Home-made string-driven table.

adopted by another friend who poured a cement base for his machine. This is how it was done:

The paper template which came with the machine was fastened to the smooth side of a sheet of Masonite and the outlines traced in red pencil. Some 2 by 1 by 3-inch pieces of wood were cut and nailed here and there along the penciled line, being kept to the *inside,* the portion where the opening would be. A long strip of cardboard (not corrugated) was cut about 2 in. wide, or the height of the sticks, and stapled to them with its bottom edge flush with the Masonite. This strip of cardboard followed the pencil line all the way around to form the inside contour of the mold. Wooden dowels of the proper size were screwed to the Masonite in places where the mounting bolts would go, and the ouside edges were enclosed with laths to the exact size of the support installation. Next a piece of coarse wire, something between window screening and chicken wire, was cut to the template shape and the outer edges were bent over, using a pair of wooden blocks. The bent-over portions were trimmed to half the height of the slab thickness so that the wire would be in the middle of the concrete, to act as a reinforcement, when it was poured. At this point, the bottom of the cardboard was sealed with modeling clay (putty would do as well) to prevent any leaks when the cement was poured, and the wooden pegs for the mounting bolts were soaked in water to make them swell slightly. The cement, a 4-to-1 mixture, was poured to a depth of 1 in. and the Masonite sheet was shaken about for a few minutes to release any trapped air bubbles. The screen was then put in place and the rest of the cement poured. A further bit of shaking and the slab was left to harden. To prevent cracking, the raw cement was sprinkled with water twice a day for 3 days before the forms were removed and the dowels driven out. Should a cement slab in view be aesthetically offensive it can be disguised with paint or veneer, or covered with one of the new papers which adhere with glue and are figured to imitate leather, marble, wood, cloth, and other substances.

Such a turntable base, rigidly supported, not only damps out mechanical noises in the machine itself, but is proof against acoustic feedback no matter what the relative positions of machine and speakers.

Mass Used in the Turntable

The other method of eliminating noises with mass is the one commonly used by the manufactures of turntables; that is, making the turntable itself heavy. One maker has a machine with a 28-pound steel turntable which he claims has a noise level of —70 db.

A heavy turntable, in addition to damping out its own bearing noises, turns with such a powerful flywheel action that motor eccentricities can affect it very little. Hence, an inexpensive motor may be used.

This method is the one I used in construction of the turntable shown in Figs. 2 and 3. The machine is almost absolutely silent, turns at a constant speed with no hint of wow, flutter, or drift, was fun to build, and cost only $21.20. Following is a parts list:

¾-inch aluminum plate, 13 in. square at 40¢ per lb. for non-ferrous scrap	$5.00
Four-pole motor bought at a "war surplus" store	2.00
One 7-inch length of ⅝-inch drill rod	.60
2 pillow blocks	2.00
One 5-inch aluminum pulley	1.50
One ⅝-inch steel ball	gift
4 ball-type furniture casters	2.00
8 ft. chalk-line	.10
One switch (SPST)	.10
One 0.05-μfd 400-volt capacitor to eliminate switch pop	.10
¾ sheet of birch ½-inch plywood	6.00
Registration fee for Evening High School Machine Shop Class	1.80
Total	$21.20

A duplicate of this turntable may cost the average reader either more or less, depending upon locality, availability of parts, and his willingness to do the work himself, assuming that machines are accessible to him and he has the ability to operate them. However, even with the machining hired done, the total should not rise to more than about $40, or what a fairly good changer would cost and considerably less than the price of a deluxe commercial turntable.

Turntable Design

The drive chosen was that using a string between the motor pulley and the turntable. Its advantages are several: there are no rubber idlers or drive wheels to wear out or develop audible flats; the string, being flexible, has enough give to absorb most noises before they reach the turntable; the turntable itself is heavy enough (18 lb.), and its inertia great enough, to damp out any eccentricities that *do* get through.

To provide additional insurance against motor noise, the motor is physically separated from the turntable support yet enclosed within it. This was accomplished by housing the motor in its own box supported on 4 furniture casters inside the turntable base, the only contact between the two being 3 springs. Besides maintaining constant tension on the string drive, the springs damp out motor vibration transferred from the motor support box to the base.

In my bachelor home this assembly sits on the floor. If it is necessary to place it in cabinetry, the only precaution to be observed is to isolate the bottom of the turntable base, possibly with rubber feet or strips of foam rubber, from the surface which supports the motor box.

Fig. 3. Motor and pillow-block mounts.

Only one speed, 33⅓ rpm, is provided for in this design; my record collection consists only of LP's and 78's. The latter are played on a changer. If you want to have other speeds available it will be necessary to use either a stepped motor pulley or separate pulleys for each speed.

The turntable itself is aluminum.

This metal was chosen rather than steel because it exerts no drag on magnetic cartridge coils. Its shaft is ⅝-inch drill rod turned absolutely flat on the bottom where it rests on a single ball. Two pillow blocks, with oilite inserts, provide vertical support. The record peg is a part of the shaft, having been turned at the same time as the turntable to insure concentricity. Figs. 4 and 5 are orthographic views of the general arrangement.

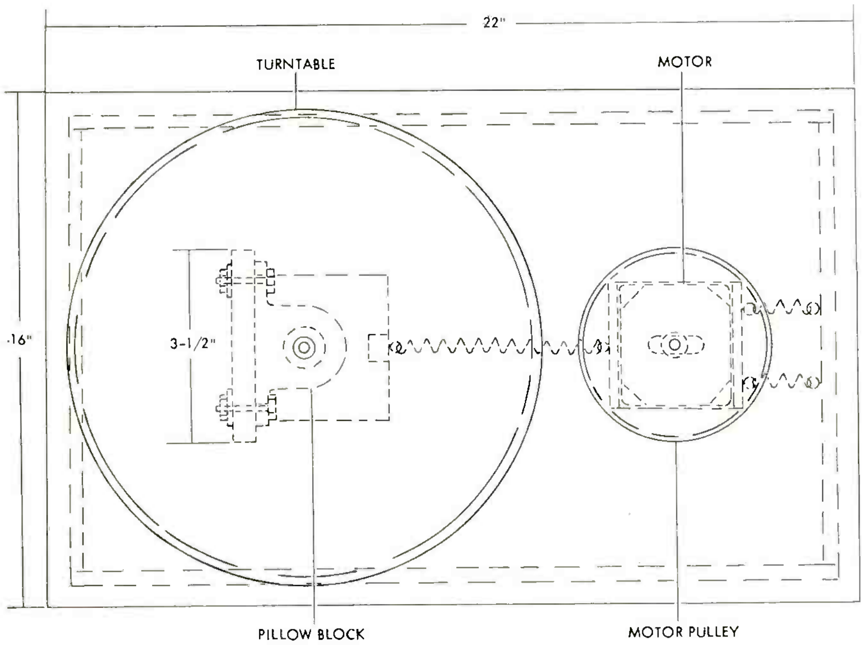


Fig. 4. Upper plan view of the turntable. Motor pulley size depends on motor speed.

Although the best available motor should be used, the design of this turntable is flexible enough so that almost any sort can be employed. Motor speed is unimportant since a pulley or pulleys will determine the speed of the turntable. A geared motor from an old 78 rpm phonograph would be usable in a design similar to this; my motor turns, under load, at an unusually slow 86 rpm. A fast motor — that is, one turning at any speed from 1,300 to 1,800 rpm — requires a separate step-down countershaft pulley and two string belts. This complicates things only slightly. Fig. 6 shows such an arrangement with the countershaft pulley base rubber-mounted for quietness. It should be noted that in such a case the springs must be so arranged that they exert pull away from the countershaft pulley, rather than from the turntable shaft.

Should you find that your motor turns in the wrong direction, you can either reverse the field connections (on an induction motor) or cross the string drive. On some motors, leads can be switched for the same result.

Many motors, particularly those found in junk shops or "war surplus" stores, are not labeled as to speed. The method of determining a motor's true speed under load is quite simple. When the turntable has been built and installed in its supports, the motor is mounted in any way convenient with a piece of adhesive tape wrapped around its shaft. The tape is used only to provide traction for the string which, by the way, need not be spliced — a simple knotted loop will do for this test. Next, a broom straw or any similar object is Scotch-taped to the surface of the turntable in such a way that one end projects over the edge far enough to strike against the arm base as the table revolves, giving an audible signal on each revolution. Following a sweep-second hand with the eyes, it is easy to get an accurate count of the rpm. Three counts of one minute's duration each should be averaged for complete accuracy. Three things will then be known: the number of turntable rpm, the size of the motor shaft diameter (check with micrometer, and include the tape), and the diameter of the turntable groove bottom. Multiplying the diameter of the turntable groove bottom by the turntable rpm and dividing the result by the diameter of the motor shaft will give true motor speed.

This known, it is a matter of simple algebra to determine the proper size for the motor pulley (and countershaft pulleys, if a high-speed motor is to be used).

In the case of a slow motor and only one pulley, the formula is: motor pulley groove diameter is to turntable groove diameter as turntable speed is to motor speed. In my case this was $\frac{x}{12} = \frac{33.33}{86}$, giving a motor pulley groove diameter of 4.65 in.

When a fast motor and countershaft pulleys are used, the formula can be expressed as

$$\frac{D_m \times D_2}{D_t \times D_1} = \frac{S_t}{S_m},$$

where: D_m is the motor pulley groove diameter.
D_t is the turntable groove diameter.
D_1 is the groove diameter of the countershaft pulley belted to motor.
D_2 is the groove diameter of the countershaft pulley belted to turntable.
S_t is the turntable speed in rpm.
S_m is the motor speed in rpm.

Example: In the case of a common motor speed of 1725 rpm, using a 12½-inch turntable with a 12-inch groove bottom diameter, a motor pulley groove bottom diameter of 1.20 in., and an arbitrary 6-inch groove diameter for the pulley belted to the motor, we wish to find the proper diameter for the pulley belted to the turntable. Thus:

$$\frac{1.20 \times D_2}{12 \times 6} = \frac{33.33}{1725}$$

$$\frac{1.20\ D_2}{72} = \frac{33.33}{1725}$$

$$2070\ D_2 = 2399.8$$

$$D_2 = \frac{2399.8}{2070} = 1.159 \text{ in.}$$

As can be seen, any of these figures can be varied as long as the proportions are maintained, thus allowing considerable latitude in pulley sizes and motor speeds. Perhaps the simplest procedure is to cut the turntable and motor pulley

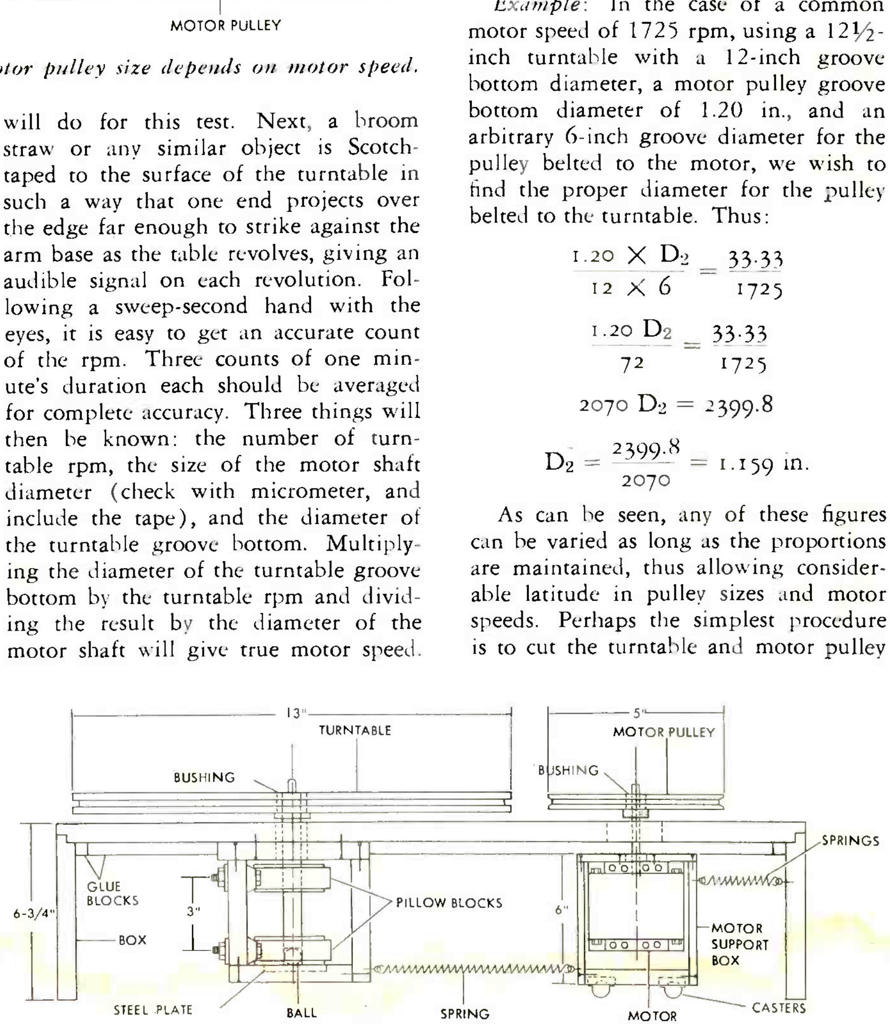


Fig. 5. Side view of turntable. Note how pillow blocks are used to hold shaft bearings.

gro… l sizes, and … pulley groo… ntable speed.

Construction

The details of machining the turntable and pulley are given both as a guide for those who wish to do their own work and to point up some of the problems which arise in trying to turn such shapes.

The ¾-inch aluminum slab was originally about 13 in. square. The corners were sawed off to lessen lathe time and the slab mounted in a 4-jawed chuck for roughing. Since the outer edges had to be turned, but at the same time held by the chuck jaws, half the thickness was turned at one time. Then the slab was reversed with the jaws holding the already-machined edge, while the other half was cut and the hole for the shaft drilled and bored out. Next, a steel bushing with a collar was made for the hole in the turntable center. (Originally this was done simply because I made a mistake and bored out the hole too large but, since it provided additional and needed contact with the shaft, it was left in the design.)

The length of drill rod was held in a collet and its butt faced off square, leaving neither an indentation nor a projection in the center. The rod was then pressed through the bushing in the turntable until the rough end extended about 1¼ in. beyond the top face. This section of rod was to serve as a support while the groove was turned and the

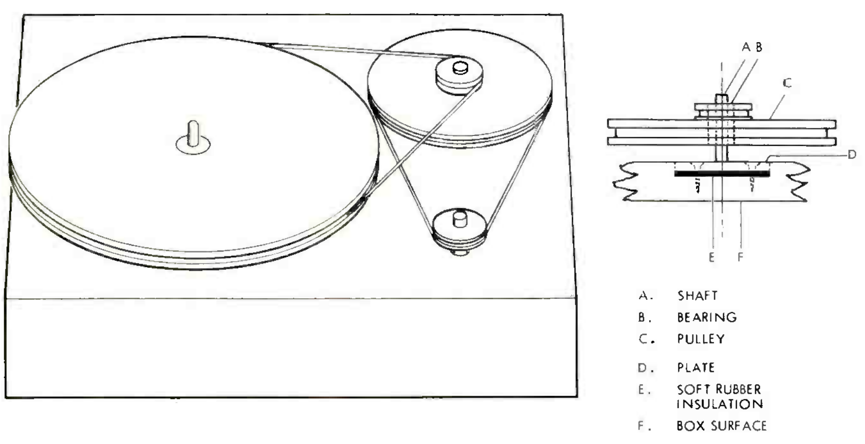


Fig. 6. Dual-string drive with countershaft pulley for use with high-speed motors.

finish cuts made and, when this was finished, was turned to size for the record peg. By drilling a center hole in the short stub to accomodate the tail-stock center, the turntable could be moved out from the collet so that the cutting tool could reach the underside for the final cut. When all turntable cuts were finished, the shaft was slipped back into the collet; the stub was then cut off ¾ in. and turned to 0.282 in., the proper size for record holes.

the difficu… n exerting enough pressure on th… to prevent slippage, especially while machining the flat-bottomed groove, leads to the belief that a larger shaft — say, 1 or 1¼ in. — would be more suitable. Collets, of course, must be used rather than a chuck because the shaft cannot be marred by tooth marks.

The bushing for the motor shaft and pulley was turned next, leaving enough collar space for a set screw. Since the height of the pulley on the shaft had to be made adjustable, a slip-fit was indicated for the center hole while the outside diameter was turned for a press-fit. After the bushing was pressed, the pulley was machined, being held in the chuck by the bushing collar because the small size of the motor shaft (5/16 in.) did not make turning the pulley feasible while it was attached. Although absolute concentricity had to be sacrificed here, the error was so small as to make no difference in the final performance.

Machining a countershaft pulley would best be done on an expanding mandrel after the center hole had been bored out for a press fit with the outer ball-bearing race.

The wooden parts present no difficulty since they are essentially nothing but 2 boxes with a few pieces of wood inside the big one to support the pillow blocks. The large box was made with mitred edges and glue blocks around the top inside corners, so that no nails had to be driven from the outside; since it was of birch it was finished blond, using several coats of clear lacquer, each lightly sanded before the next was applied.

The little open-sided box which holds the motor was simply tacked together from a few pieces of scrap wood, and 4 furniture casters were then nailed to its bottom. Screw eyes were added for the 3 springs.

Only one precaution must be observed when making the large box: keep everything square, since the pillow blocks which support the shaft must not bind at any point and the turntable must be level in the final installation. In fact, although it would entail more work and additional expense, for those seeking perfection and greater ease of assembly I recommend a shaft support such as shown in Fig. 7. This can be made by welding (or threading and screwing) a mounting flange to the end of a 1¼-inch length of hot-rolled steel. This is for a ⅝-inch shaft; use 1¾ in. for a 1-inch shaft. Then drill and bore it out to fit, leaving 2 surfaces inside which will be finished to exact size with a tool-post grinder. Finally, true the flange face.

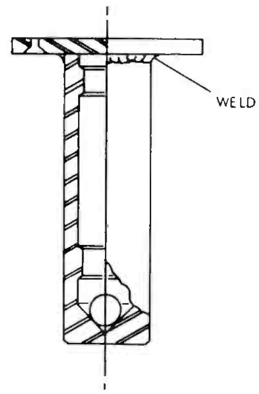


Fig. 7. Alternative shaft bearing well.

With this sort of well to support the turntable a smaller ball can be used, providing a smaller point of contact with the shaft butt. Also, the well can be filled with oil for added shaft life. This is, in fact, the sort of support used by commercial turntable manufacturers.

Either the Long or Transmission splice* may be used to put the belts together from 3- or 4-strand chalk line. I prefer the former as being easier to tie, although both require a little patience. The endless belts used on dental drills would be ideal in an installation where the motor pulley and turntable are at some distance from each other, because the shortest length made is 114 in.

It is also possible that neoprene tubing can be chemically welded to form belts. However, I have found spliced chalk line to be very satisfactory since, if properly tied, the splice is no larger than the rest of the string.

A flat belt could be employed satisfactorily for direct turntable drive with a motor which ran at fairly high speed, by using a turntable whose outer edge was turned slightly convex rather than grooved. Such a belt, running directly from a small-diameter motor pulley to the turntable, could easily be sewn together to the length needed. There would be no danger of the belt riding off the turntable edge provided the shaft and motor spindle were kept parallel with each other, and the belt perpendicular to both.

*Ashley, Clifford W.: **The Ashley Book of Knots**, Doubleday, 1945.

Loa [illegible]

by Ernest B. Schoedsack

LOAD it with sand! No, I don't mean that you should substitute a load of sand for the air loading of your horn or cabinet. Sand loading refers only to an excellent method of damping your speaker panel, or, if you don't mind the weight and work, all the panels of your speaker enclosure. In so doing you are almost certain to make some improvement in the range and smoothness of your speaker response, and quite likely to make a really worthwhile difference in the performance of your installation.

If you are skeptical, try the following simple experiment: rap your knuckles sharply on a ¾-inch plywood board. You will hear a loud, sharp, ringing sound. If you double the thickness of the wood by screwing and gluing another board to the first, and rap again, you will still produce a ringing noise, only slightly duller and of lower pitch. An electric vibrator, or an oscillating sander, held firmly against one side of the board can be felt distinctly on the other side.

Now let us take two thin pieces of plywood, say ½ or ⅜ in. thick, of similar dimensions. Space them ¾ in. apart with wooden strips to form a hollow panel, and fill the space with clean, dry sand. Now repeat the rapping and application of the vibrator. The result of the rap will be a faint, dull, low-pitched sound, while the chances are that the vibrator cannot be felt at all on the other side of the panel.

The reason for this phenomenon is that all wood which is hard and strong enough for cabinet construction is more or less resonant; it will transmit vibration applied to it, and will set up vibrations of its own, which in turn are communicated to the air, forming air waves or sound.

Sand, however, practically inert due to its weight and density combined with its non-cohesion, is as nearly the perfect foil for vibration as any material can be.

This fact has for the audiophile a very useful application. Inasmuch as every physical action has an equal reaction, the vibrations of a loudspeaker cone must push as hard against the speaker chassis as they do against the air. Therefore, if the baffle board to which the loudspeaker is secured is not inert, the speaker will set up vibrations in the board itself. These will of course vary greatly, depending on the power and frequency of the vibrations, and the size and resonant qualities of the wood.

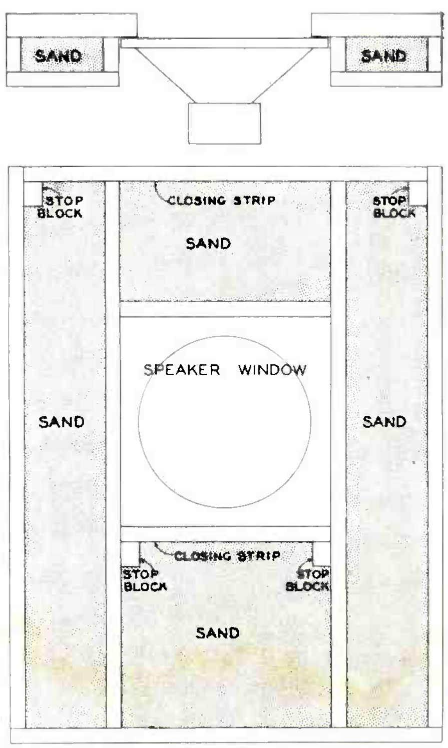
When this happens, at least two very undesirable things occur. First, energy that the speaker cone should be transmitting to the air is being dissipated into the structure of the cabinet. Now from the standpoint of loss of general efficiency this *should* not be very important, but it is, because the vibrations or sound waves most readily lost to the wood are naturally the heavier, longer waves — i.e., the bass tones. Since the chief purpose of the baffle or enclosure is to reinforce and extend the bass frequencies, a vibrating baffle tends to defeat its own objective.

Second, a good cone is carefully engineered to reproduce faithfully the electronic signals sent to it from the amplifier, and not to *add* sounds of its own manufacture; aided by a good amplifier, it is supposed to check the oscillations caused by each sound instantly as the electrical signal is finished.

Now when random vibrations, no matter how slight, are allowed to run through the wood of the cabinet, uncontrolled air waves are produced, and that old devil distortion begins to raise its ugly head. Overhanging loud notes begin to jumble each other because of lack of panel damping.

No speaker has an absolutely flat response; just look at the curves published by the best speaker manufacturers. They look like seismographic charts of an earthquake, but in a proper enclosure and room these sharp-looking peaks and valleys usually pass unnoticed in the music. But a vibrating speaker board, if its principle response is in phase with one of these peaks, can cause most unpleasant effects by picking up and multiplying this fault.

Fig. 1. Sand-filled panel sectional views.



All the foregoing disastrous effects of vibrating baffle boards vary greatly, of course, with the quality and construction of the enclosure, but if you think your enclosure is immune from such losses and distortion, just try one more little experiment.

Put on a record which includes wide range and plenty of bass. Turn it up to a good full volume and hold your hand on the speaker board and various parts of the cabinet. If you feel any very perceptible vibration at any time, your cabinet isn't what it ought to be, for distorted air waves are being produced. Even if they do not stand out as distinctly unpleasant tones, they are mixing into your music and doing it no good. If you are a perfectionist, this is something to get rid of.

The remedy — sand loading — is quite cheap and simple, and you may be amazed at the improvement in general cleanness of sound and the extension and reinforcement of the bass.

Sand loading of baffle boards or cabinet panels should not be confused with insulation by means of felt, Fiberglas, or rock-wool. The purpose of insulation is to absorb middle- and high-frequency sound waves, and to prevent their reflection and cancellation, within the enclosure. It cannot prevent the vibration of the board to which the speaker is bolted. Only a vibration-proof board can do that.

The sand loading idea is not mine. It originated, so far as I know, with G. A. Briggs of the Wharfedale Wireless Works in Great Britain — a gentleman wise in the ways of sound. He has designed and produced a successful sand loaded baffle. Since hearing this device, I have done a little experimenting of my own with a view to finding a substitute for sand, which in a layer of an inch or so makes a panel very heavy. Diatomaceous earth is an excellent heat insulator, and as heat insulators are often good sound insulators, it was thought

Continued on page 41

by E. B. Mullings

Using Test Instruments

Vacuum-Tube Voltmeters, Part I

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

If you are sufficiently curious about electronic devices to look behind the front panel or under the chassis, and wonder what makes them work — you are well on your way to being a test equipment user. Or perhaps you already have some instruments of your own, and have been using them. Fine! The best way to learn is by doing, so long as care is exercised until you become completely familiar with your instruments.

But whether you have been using test instruments or not up to this time, you can profit by finding out more about them and by reviewing some of the more usual applications to which they are suited. It stands to reason that the more you know about how a particular piece of test equipment works, the more you can make it do, and the more it will be worth to you. Take the VTVM (vacuum-tube voltmeter), for example. Understanding how it functions will enable you to appreciate more fully its capabilities and its practical limitations.

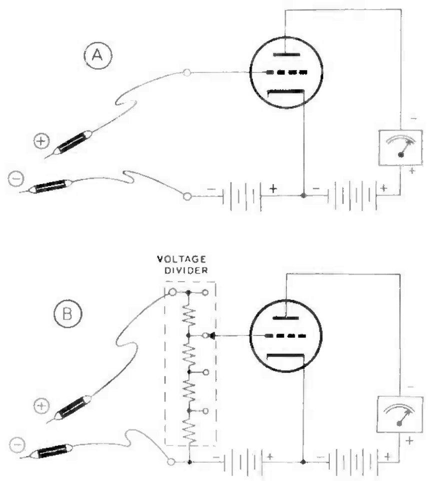
Perhaps it would be a good idea to discuss at the beginning the differences between a standard multitester and a VTVM, and why the latter should be chosen for treatment rather than the former. A multitester normally requires no AC power, so it can be used anywhere. It is often smaller and more rugged than a VTVM, and simpler to operate and maintain — and it is likely to be less expensive. Why should the amateur user bother with a VTVM?

The answer is that a VTVM is much more versatile, and is actually easier to use than a multitester. Because current to operate the meter movement in a multitester must be drawn from the circuit being tested, it happens very often that normal operation of the circuit is disturbed by the meter. This leads to inaccurate readings. A VTVM does not have this limitation, so unless you know a good deal about circuit theory you can be a lot more confident in readings obtained with a VTVM. Furthermore, since a VTVM provides amplification, it can normally accommodate a greater range of input voltages and currents.

Because of their portability and convenience, multitesters are used by repairmen on outside calls, but VTVM's are used for shop work. If you can afford only one, then it makes sense to get the more flexible unit.

Basic DC VTVM Circuit

The vacuum tube voltmeter can be thought of simply as a triode amplifier stage in which an unknown voltage is applied to the control grid of the tube, producing a change in the plate current. The plate current is registered on a meter. Such a simplified VTVM circuit is shown in Fig. 1A. The control grid voltage in this amplifier determines the amount of electron flow from cathode to plate and therefore the amount of plate current flowing through the meter. Positive grid voltage results in increased plate current, while negative grid voltage results in decreased plate current. By applying a fixed negative (bias) voltage



Figs. 1A, above; 1B, below. Basic VTVM.

to the control grid, the tube is biased to cutoff (the point at which plate current ceases to flow) and the meter reads zero. The voltage to be measured is then applied to the test leads with a polarity opposite to that of the bias voltage, canceling part of the bias and allowing the plate current to flow and the meter to read. In other words, if 3 volts negative bias were applied, cutting the tube off, and then 1 volt positive were applied to the test leads, the bias would be reduced from 3 volts to 2 volts and the meter would read up-scale accordingly. Or, if a positive 2 volts were applied at the test leads, the negative bias would be canceled back to 1 volt: the plate current flow, and accordingly the meter reading, would be even higher. Once the meter were calibrated in terms of the grid voltage affecting the plate current, DC voltage from zero to 3 volts could be measured with this simplified VTVM.

Two characteristics of such an instrument are worthy of note at this point. First, the markings on the meter scale would be nonlinear — that is, not spaced equally across the meter face. The vacuum tube itself is responsible for this, because the relationship between grid voltage and resulting plate current is not linear. More about this later.

Second, note that the range of such a VTVM would be limited to measurements between zero and 3 volts DC, hardly satisfactory for practical use. But this can be remedied with little difficulty.

Extending the Range

Fig. 1B shows how a voltage divider is used in a VTVM circuit to increase its range. The voltage to be measured is applied across the entire voltage divider network, while only part of the total voltage is actually connected to the control grid of the tube. In this way, the grid voltage can swing between zero and 3 volts while the VTVM is actually measuring much higher voltages — for instance, zero to 10, 30, 100, or more volts, depending on what percentage of the applied voltage is passed to the grid. The appropriate scale on the meter face is read, according to where the range switch of the voltage divider is set for the measurement in question.

The VTVM represented by this simplified circuit has high input impedance — that is, the voltage divider and the grid circuit are both large resistances and, therefore, have very little effect on the circuit being tested. Too, actual gain occurs in the triode circuit, since it takes only a relatively small change in grid voltage to effect a large change in the meter reading. The vacuum tube serves further to isolate the voltage under test from the meter movement. There is no direct connection between the two, and this affords protection to the meter because above a certain grid voltage there is no further increase in plate current.

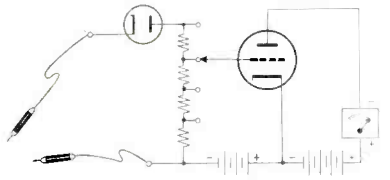


Fig. 2. Diode rectifier makes AC VTVM.

These are all desirable characteristics in a measuring instrument.

A VTVM can do more than measure DC voltage, however, and this basic circuit can be extended in function to measure AC voltage and resistance with very little trouble. This is desirable too, in order to make it a multipurpose device.

AC VTVM Circuit

Simply by adding a diode rectifier ahead of the voltage divider, as shown in Fig. 2, the DC VTVM is converted to one that will measure AC. The diode rectifies the AC voltage so that only pulsating DC reaches the voltage divider, vacuum tube, and meter. The meter deflection, even though actuated by rectified AC instead of pure DC, is proportional to the AC voltage applied to the test leads, and the meter can be calibrated accordingly. However, note that the meter can be calibrated in terms of AC (RMS) or AC (peak-to-peak), or both, depending on the intended use of the instrument. AC (RMS) is the most common calibration used, since it is the effective equivalent of an equal quantity of DC. In other words, 110 volts AC (RMS) will do the same amount of work as 110 volts DC. There are occasions, though, when it is important to know the peak-to-peak value of an AC voltage. This can be calculated (for sine-wave inputs) by multiplying the RMS value by 2.83, but the calculation is unnecessary if the meter scale is calibrated for peak-to-peak measurements to begin with. Further, for non-sinusoidal input waveforms, the relation between RMS and peak-to-peak values is not 2.83. Fig. 3 shows the difference between RMS and peak-to-peak values of a sine wave.

Vacuum-Tube Ohmmeter

The same simplified VTVM circuit is outlined in Fig. 4, except that the input circuit has been changed to facilitate resistance measurements. The grid of the vacuum tube still responds to DC voltage, but the additional battery and voltage divider function in such a way that the voltage reaching the grid is inversely proportional to the amount of resistance connected between the test leads. When this is true the meter can be calibrated in ohms, and the VTVM will measure resistance directly.

A string of resistors is connected in series with a battery across the grid circuit. Since no current can flow in the grid circuit (because the circuit is effectively open between cathode and grid) there will be no voltage drop across the resistors; the battery voltage, acting against the bias voltage, causes the meter to deflect up scale. This is a shunt-type ohmmeter, in which the normal maximum resistance reading (infinity) is at the right end of the meter. At the other extreme, when the test leads are shorted together, a circuit is completed around through the test leads, placing the resistors and the battery in series with current flowing. Battery voltage is dropped across the resistors, and the grid voltage returns to the normal bias level (cutoff). The meter then reads zero. But when an unknown resistance is connected between the test leads, the amount of voltage dropped by the voltage divider resistors depends on the value of the

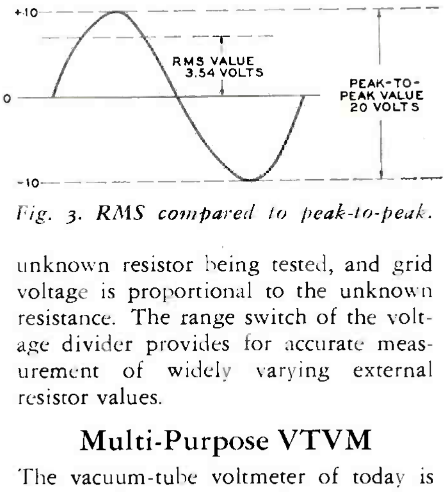


Fig. 3. RMS compared to peak-to-peak.

unknown resistor being tested, and grid voltage is proportional to the unknown resistance. The range switch of the voltage divider provides for accurate measurement of widely varying external resistor values.

Multi-Purpose VTVM

The vacuum-tube voltmeter of today is only a refined version of the basic circuits described above. These simplified circuits are valuable in thinking about the basic principles upon which the VTVM functions — because the basic principles have not changed. The refinements that make the modern VTVM a practical, accurate, and useful instrument tend to obscure the basic circuits. They are still there though, and, by thinking them through, the more complex finished circuit can be clarified and simplified in your mind. A schematic of a modern precision VTVM is given in Fig. 5. It is different from the simplified diagrams in many ways, but for good reasons. Here are some of the reasons.

Probably the most significant difference between the simplified circuits and the modern VTVM circuit is the use of a twin-triode tube rather than a single triode. This offers several advantages in instrument performance. The meter of the VTVM is connected between the cathodes of 2 triode stages so that it measures the potential difference between the 2 cathode resistances. A voltage to be tested is applied to the control grid of one triode section only. Note that the plates of the 2 tube sections are connected together so that they are common to both sections. The effect of this is to balance one tube against the other, thereby canceling the effects of line voltage or power supply variations, and the effects of the tube characteristic curve. When working one tube against an identical tube, the relationship between the test voltage applied to the grid, and the meter indicating current, is *linear*, and the meter can be calibrated with a linear scale. Remember that nonlinearity was one of the characteristics of the single-triode VTVM circuit. Linear operation and calibration are decided advantages.

Of course, a practical VTVM circuit requires extra controls to balance and calibrate the instrument initially and to assure accuracy for all functions. Inspection of the schematic in Fig. 5 will reveal a zero adjust control that sets up the initial balance between the 2 triode sections of the 12AU7 tube, a DC calibrate potentiometer, an AC calibrate potentiometer, and an ohms adjustment for setting the meter pointer to the proper point on the scale for resistance measurements. Also, in order to keep from having too many switches and controls on the front panel, and to simplify operation, a function switch must be provided to rearrange the circuit for DC voltage measurements, AC voltage measurements, or resistance measurements. This switches the diode tube in or out for AC voltage tests and connects the battery and voltage divider network into the circuit for resistance measurements. The resistance voltage divider and the divider used for voltage tests are ganged on the same range switch so that either

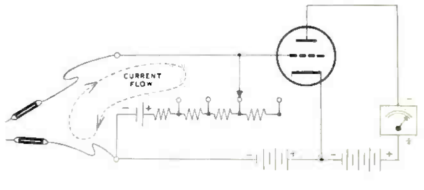
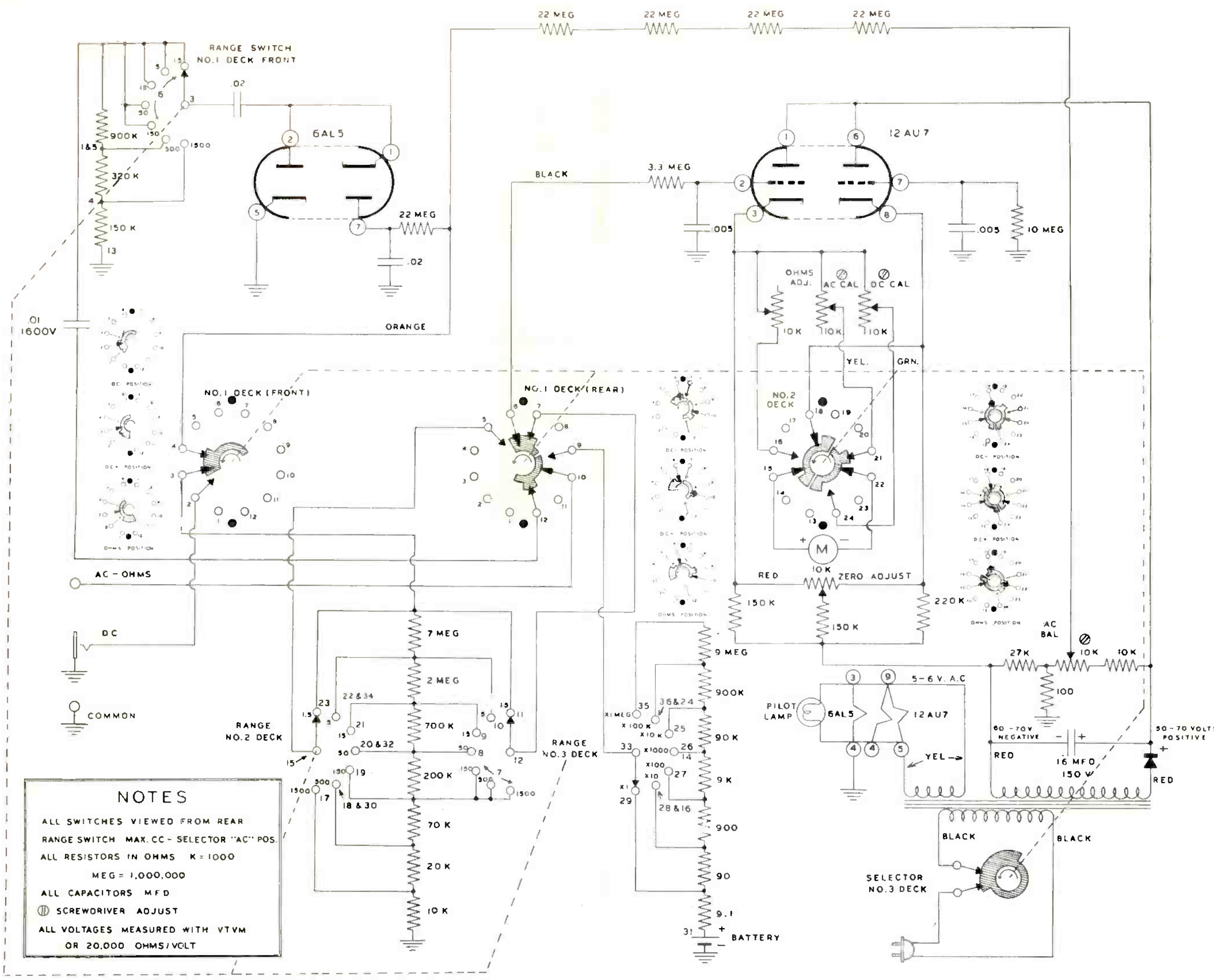


Fig. 4. Circuit converted to an ohmmeter.

resistance or voltage ranges may be set up with one rotary switch.

Instead of using batteries for plate and bias voltage as in the simplified diagrams, the modern VTVM ordinarily has a conventional power supply that furnishes DC voltages from the standard AC power line.

Look over the schematic diagram in Fig. 5 carefully, and trace out the circuits for various VTVM functions. You



COURTESY HEATH COMPANY

Fig. 5. Complete circuit diagram for commercial VTVM, with facilities for measuring ohms and AC or DC volts. Picture is below.

will find that they closely parallel the basic circuits described previously.

Instrument Accuracy

Assuming a reliable source for calibration, the accuracy of a modern multi-purpose VTVM depends primarily upon the accuracy of the meter itself and the resistors used in the voltage divider networks. The voltage divider resistors must be held to close tolerance because they play such a large part in instrument operation. For the VTVM shown in Fig. 5 the overall accuracy is specified at 3% of full-scale reading. For most accuracy, then, the user should set the range switch to obtain nearest full-scale meter indication.

The 3% tolerance can be broken down into a 2% allowance for the meter movement itself, and a 1% tolerance for the precision voltage-divider resistors. Since any situation of this nature seldom results in all tolerances being additive, it is more likely that the overall accuracy of the vacuum tube voltmeter would be something better than 2%.

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Controls and Operation

The front panel of this typical VTVM has only 4 controls, as can be seen in the photograph. Control functions are consolidated for simplified operation. One rotary switch provides function selection so the instrument can be switched from AC to DC or resistance measurements, while another rotary switch provides for range selection. The same switch is used for AC, DC, and resistance ranges. In addition there are a zero adjust control and an ohms adjust control. Various calibrating and balance controls are inside the instrument case, since they need adjustment only rarely. Rather than having separate pairs of test leads for different types of measurements, the instrument requires only 3 test leads. One is the common lead for all types of measurements, one is used with the common lead for DC measurements only, and the other is used with the common lead for AC and resistance measurements. The test leads plug into receptacles on the front panel.

Understanding the internal workings of a VTVM is the first step to finding out what it is capable of doing for you. The second step is to learn some of the specific tests that can be made with it, and see exactly how it should be used to best advantage in various applications. We will turn our attention to this subject in a forthcoming issue.

How They Did It

BINDING A TAPE LIBRARY

by John Hoke

Tape albums can be as attractive and durable as record jackets.

First step is to make a hinge or to reinforce the existing one.

NOW that tape recorders are well within the financial and technical reach of the average audio enthusiast, disc records can be preserved indefinitely by transcribing them to tape. Valuable old records can be re-recorded on tape and the breaks between records in an album can be edited out, to make a continuous unbroken performance. Not to be overlooked either is the possibility of actually improving on disc playback quality by means of filters, editing of pops, and so on. Further, pre-recorded tapes are becoming available in ever-increasing quantity. All this means that tape libraries can and are being accumulated by many users.

Disc albums that include a great many records often require the use of several reels of tape in order to re-record the complete work. While there is space provided on the tape boxes for descriptive matter, these boxes — with hand-written material on them — are not the most attractive means of packaging the collection. There is also a constant possibility of mixup and a grave danger that some valuable recording may be erased because of unclear marking.

With a roll of 1-inch black masking tape, photographic dry-mounting tissue, a can of pressurized plastic spray, and the assistance of a photographer friend, you have the ingredients for making attractively bound albums for your tape collection. Here is how I did it.

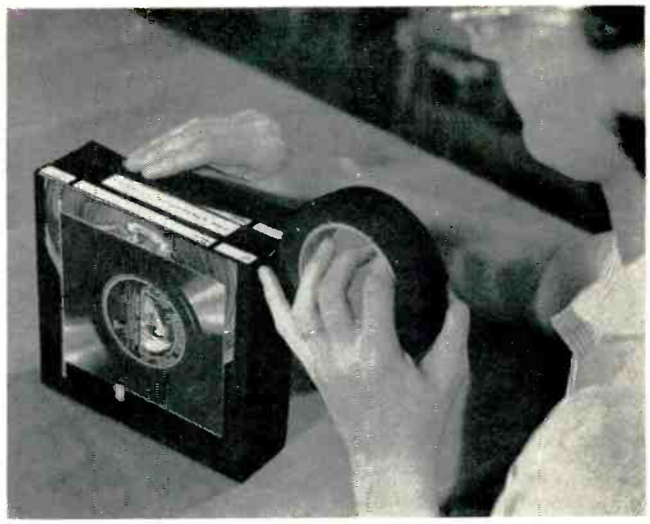
All the tape reels used to record any given work or collection are bound into a single album. Some tape reel boxes are hinged along one edge. Reinforce the inner edge of this hinge (or make a hinge) with a strip of black masking tape, running your finger down the seam carefully in order to stick it down firmly.

Reinforce the outer edges of the box

To avoid fraying box edges, reinforce them with masking tape.

Join boxes in a group by making masking-tape hinges as before.

Photograph can be dry-mounted on cover box to serve as label.



Title of collection is typed and installed on album backbone.

cover by extending a length of tape all around the three unhinged edges, running the tape parallel to the lip-edge of the box lid. Fold down the overlapping tape around the joining edge of the box lip and cover. Slit the tape where it goes around the two corners of the box, and fold it neatly down over the corner. Cut away the tape where it covers the thumb-insert used to help open the box.

When all the boxes have been similarly reinforced they are ready to be joined together. With the hinged edges of the boxes facing to the left, stack them in proper sequence with the first box on top and the last on the bottom. Open the first box as a whole away from the second, as you would the cover of a book, and lay it down edge to edge with the second box. Place a length of tape along the adjoining edges, overlapping both boxes. Follow this procedure with each box in the stack, until all boxes are hinged together by their inner edges, and will close like pages in a book.

Before completing the binding, type out the title of the recorded work or collection on a piece of bond paper. Letters of the title should read from top to bottom in the vertical row. Go over each letter several times until it is good and black. Cut out the little strip and cement it to the backbone of the middle box, centering it attractively. Put black masking tape above and below the title strip, extending the tape to the top and bottom edges of the boxes, and allow the tape to overlap the paper on which the title is printed. Now apply a strip of tape along the full edge of the backbone on each side of the title strip, again overlapping the paper slightly. Usually the two pieces of tape will be wide enough — when binding a 3-box album — to be folded over the edges of the front and back covers of the album. Trim the ends of the tape with a razor blade so that the ends do not extend over the edges of the binding job.

A cover for the album can be prepared that is not only good-looking, but which provides information about the work. Photograph the label of the first record in such a way that the title of the work, as well as other pertinent information, appears about original size. Make a print slightly smaller than the box covers, about 6 by 6 in. Affix the print to the cover with photographic dry-mounting tissue. You are now ready to complete the mounting job.

Extend 2 lengths of masking tape along the right and left sides of the cover surface, overlapping the edges of the picture as well as the binding at the edges of the box. Then, to reinforce the whole binding job, run a length of tape around the top of the backbone binding, and along the top edge of the back cover. Follow the same procedure for the bottom edge of the album. Allow a little tape on each end to fold under the box edge for added reinforcement. If there is more information you wish to include with the album, there is additional space for it on the inside covers and the back cover.

The album is then given a few coats of plastic spray to add luster and give protection to the binding, photographs, and labels.

Tape is run along top and bottom edges to strengthen binding.

Complete the job by applying several coats of a plastic spray.

BASIC ELECTRONICS

II: Volts and Amperes

by Roy F. Allison

IN Chapter I, it will be recalled, we discussed (among other matters) the attraction of charged bodies. Little or nothing was said about the magnitude of these charges, however, and this now becomes important. First, it will be necessary to state a few definitions in the cgs (centimeter-gram-second) system of units:

A *dyne* is a measure of force; it is that force which, acting on a mass of 1 gram, causes it to be accelerated at the rate of 1 centimeter per second per second. In more familiar terms, 445,000 dynes are equivalent to 1 pound.

A *centimeter* (cm.) is a measure of length; it is equivalent to 0.3937 in.; there are 2.54 cm. per in.

A *gram* is a measure of mass and, practically speaking, of weight also; an ounce equals 28.35 grams. Hence a gram is equivalent to 980 dynes.

An *erg* is the basic unit of work, the product of force acting through a distance. It is defined as the work done when a force of 1 dyne is exerted upon a body through a distance of 1 cm.

A *joule* is a more practical measure of work; it is equal to 10 million (10^7) ergs. Related to English measurements, 1 foot-pound (ft.-lb.) equals 1.3549 joules.

Now, back to electric charge. The force between 2 charged bodies is directly proportional to the product of the charges and inversely proportional to the square of the distance between them. Stated formulatically,

$$F = \frac{Q_1 Q_2}{er^2}$$

where Q_1 and Q_2 are expressed in electrostatic units of charge, and their separation in cm. is r. F is the force in dynes. e is a constant determined by the medium surrounding the charges. For a vacuum e is 1; for air it is very close to 1, and can be neglected in our considerations. The electrostatic unit of charge, then, is a quantity of electricity which will be acted upon by a force of 1 dyne when placed in a vacuum 1 cm. from an equal quantity of electricity. The force may be one of attraction or repulsion, depending on whether the charges are opposite or similar in sign. The electrostatic unit charge, sometimes called electrostatic unit (esu) or unit charge, should not be confused with the electron charge. A unit charge consists of an excess (negative charge) or deficiency (positive charge) of 2.083 billion electrons.

It is customary to consider the region around a charged body as an electric, or electrostatic, field. Any other charge placed in this field will be attracted or repelled, and the strength of the force acting on it will vary, of course, according to the field strength. Field strength is dependent on the distance from the charge creating the field and the strength of the charge. Designated E, field strength of an electric field at any point is the force in dynes exerted on a unit charge placed at that point. This force is $F = EQ$; force in dynes is the product of field strength times the quantity of the charge in esu. Field strength at any point from a charge is, correspondingly,

$$E = \frac{Q}{r^2}$$

Here it may be a good idea to re-read the definitions for the erg and joule. Work is done whenever force is exerted on a body through some distance. Since an electric field exerts a force on a charge within the field, work must be done if the charge affected is moved toward or away from the charge creating the field. When this is done the affected charge has moved from one value of field strength to another; it has moved from one level of field intensity to a different level. Because the force on the charge has changed, the *potential* work possible with further movement has changed also. The idea of "potential" is an important one: the potential at any point is the work necessary to bring a unit charge from an infinite distance up to that point, or to remove it from that point to an infinite distance. Potential due to a charge of Q esu at a point r cm. from the charge is

$$V = \frac{Q}{r}$$

The potential V is in ergs per esu.

More important than absolute potential in practical electronics, however, is the *difference* in potential between 2 points. This can be found simply by subtracting the potential at one point from that at the other point. The work done in moving 1 esu between 2 points is equivalent to the potential difference between the points, or $W = QV$. W is the work in ergs; Q is the charge in esu; V is the potential difference in ergs per esu. Remember that work is done only when electricity moves between points for which there is a potential difference. If it flows in a wire that offers it little or no resisting force, little or no work is done and there cannot be any appreciable difference in potential along the wire.

In dealing with actual electrical measurements, the esu and the erg per esu are impractically small. Much easier to work with are the derivative units, the coulomb and the volt, respectively. A coulomb is 3 billion esu. Since an esu is 2.083 billion electron charges or electrons, then a coulomb is a charge equal to 6.25 billion billion (6.25×10^{18}) electrons. This is a *quantity* of electricity. *Current* is defined as the rate of flow of electric charge through a circuit, and it is measured in amperes. One ampere (amp.) is a current of 1 coulomb per second:

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

where I is in amps., Q is in coulombs, and t is in seconds.

The working unit of potential difference, whether in an electric field or in an electric circuit, is the volt. Potential differences exist in electrical circuits and in sources of electricity, and work must be done in moving electric charges from one potential to another here also: this will be dealt with in more detail later. The potential difference between any 2 points is 1 volt when 1 joule is expended in moving 1 coulomb of charge from one point to the other. Since 1 joule is 10 million ergs, and 1 coulomb is 3 billion esu, it follows that 1 erg per esu is equal to 300 volts. Potential difference is often referred to as "voltage".

A useful analogy to electrical quantities is found in hydraulics. Total gallons of water pumped through a closed system can be compared to the circulating electric charge in coulombs; gallons of water pumped per second is analogous to coulombs per second or amperes; pump pressure is roughly similar to potential difference or voltage.

Chemical Ionization

It will be recalled that when a chemical combination takes place such as that of sodium and chlorine, there is an actual transfer of one or more electrons from each atom of one substance to each atom or atoms of the other substance. In this process ions are formed: an ion, in this sense, is an atom having an excess or deficiency of electrons compared to its normal quantity, so that it possesses an electrical charge. In a larger sense an ion is any discrete combination of one or more atoms that possesses an electrical charge.

Ionic compounds such as sodium chloride are known as salts. In their pure states salts do not consist of individual and independent molecules (a molecule is the smallest division of a compound — for sodium chloride it would be a combination of a sodium atom and a chlorine atom). Instead, the ions are arranged in (relatively) gigantic lattice networks or crystals, with each ion of chlorine surrounded by 6 ions of sodium, and vice versa. Fig. 3 in Chapter 1 shows the ion configuration only — actually, the lattice network is made up of many billions of ions. Each grain of table salt is such a crystal. The ions are grouped rigidly together by their mutual electrostatic attraction. Neither the ions themselves nor any of their electrons is free to move about within the lattice; accordingly, salts are poor conductors of electricity when in the lattice form.

But when a salt is in aqueous form (dissolved in water), the situation is quite different. The crystalline structure is broken down. Pure water is a good insulator, which is to say that it has a high dielectric constant and, when its molecules penetrate the lattice, they reduce the attraction between oppositely charged ions. The individual sodium ions (Na^+) and chlorine ions (Cl^-) are separated and free to wander in the solution.

Hydrogen, you will recall, has one electron in its elemental state, and the first electron shell is complete with 2. Therefore hydrogen enters readily into combinations with other elements that offer the opportunity for shell completion. Ordinarily it accomplishes this by sharing its electron with an atom of another element that normally lacks one or more electrons to complete its outer ring. Except when it combines directly with oxygen to form water, an acid is formed when hydrogen compounds in this manner. Examples of acids are the compound hydrogen chloride (hydrochloric acid), consisting of one atom each of hydrogen and chlorine; hydriodic acid, one atom each of hydrogen and iodine. An acid may be formed also by the combination of hydrogen with an ionic radical — a radical is a combination of atoms that in themselves do not form a compound but which often acts as a unit in forming compounds with other elements. Such an acid is H_2SO_4, sulphuric acid, formed by 2 atoms of hydrogen combined with the sulphate radical (one atom of sulphur combined with 4 of oxygen).

In aqueous solutions, acids break up into ionic form. Each atom of hydrogen combines with a water molecule to make a hydronium ion (H_3O^+). The sulphate radical has a double negative charge ($SO_4^{=}$). These ions are free to wander about as the salt ions in solution are.

Now, leaving salts and acids for a moment, assume that a solid metal such as zinc is placed in water. Zinc atoms tend to enter the water as ions (Zn^{++}), leaving their "extra" shell electrons on the metal bar. These electrons, of course, give the bar a negative charge, and soon the accumulated charge is high enough to prevent any more positive ions from leaving the bar.

Suppose that a zinc salt is then added to the solution. This results in a great increase in the number of positive zinc ions, which will be attracted to the negative bar. When a zinc ion reaches the bar it picks up 2 free electrons and is deposited on the bar as zinc metal. The tendency of the metal to go into solution as zinc ions is called its electrolytic solution pressure, and the tendency of the ions in solution to form the metal is called osmotic pressure. The latter is affected by the concentration or activity of the solution. *For a given strength of ionic solution,* the potential of the bar with respect to the solution reaches an equilibrium point that is determined by the metal's solution pressure. Interestingly, the electrode potential under such conditions is different for various metals; that of zinc in a solution of zinc ions is more negative than that of copper in an equivalent solution of copper ions, for instance. Even more interesting is the fact that electrode potentials correlate directly with the degree of chemical activity of the elements concerned; that is, with their readiness to form stable compounds. Following is a list of electrode potentials in order. This is known as the electromotive force series.

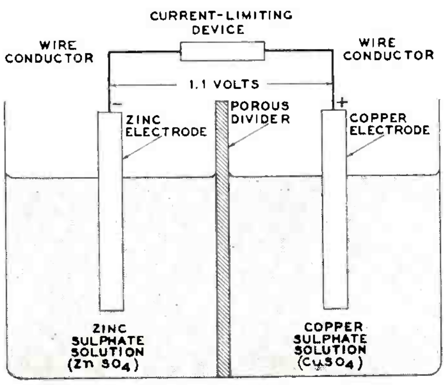


Fig. 1. Difference in electrode potential between zinc and copper produces current.

Electromotive Force Series

Electrode	Symbol	Normal Electrode Potential, Volts
Lithium	Li	—3.02
Potassium	K	—2.92
Calcium	Ca	—2.87
Sodium	Na	—2.71
Magnesium	Mg	—2.34
Aluminum	Al	—1.67
Zinc	Zn	—0.76
Iron	Fe	—0.44
Cadmium	Cd	—0.40
Nickel	Ni	—0.25
Tin	Sn	—0.13
Lead	Pb	—0.13
Hydrogen	H	0
Bismuth	Bi	+0.20
Copper	Cu	+0.34
Mercury	Hg	+0.80
Silver	Ag	+0.80
Gold	Au	+1.68

It is also significant that each element will displace those below it in the table from equivalent solutions of their salts: sodium will displace magnesium in a solution of magnesium chloride to form the compound sodium chloride. The hydrogen in acid solutions will be replaced easily, in general, by metals above hydrogen in the list; thus we say that acid "eats" iron or aluminum. This is not true of metals below hydrogen in the series — gold is extemely resistant to acid action.

Electric Cells

The phenomenon of electrode potential can be used to generate electricity by chemical means. Assume a zinc electrode suspended in a solution of zinc sulphate and a copper electrode in a solution of copper sulphate. The 2 solutions are separated by a porous plate that keeps them in contact but prevents them from mixing; see Fig. 1. The metal electrodes are connected by a wire in which an element that resists electric current to some extent is inserted.

From the electromotive force (emf) series table, we see that the electrode potential of the copper bar will be +0.34 volt and that of the zinc bar will be —0.76 volt. The zinc will then be 1.1 volts negative with respect to the copper through the external circuit. Positive copper ions in solution obtain electrons from the bar and are deposited on the bar as metal. Negative sulphate ions drift through the porous plate to the left, and positive zinc ions drift through to the right. Zinc ions continue to leave the zinc bar, leaving their electrons which make the zinc negative and drift to the top of the bar, thence to the external circuit to complete the circle. As operation continues, the zinc bar is consumed to some extent, the zinc sulphate solution becomes more concentrated, the copper bar is built up by metal deposition, and the copper sulphate solution becomes weaker and more contaminated by zinc ions.

This is a displacement cell or Daniell cell; slightly rearranged physically it is called a gravity cell. It is best used for supplying small, steady currents.

Another elementary cell is shown in Fig. 2. Here both copper and zinc electrodes are immersed in a weak sulphuric

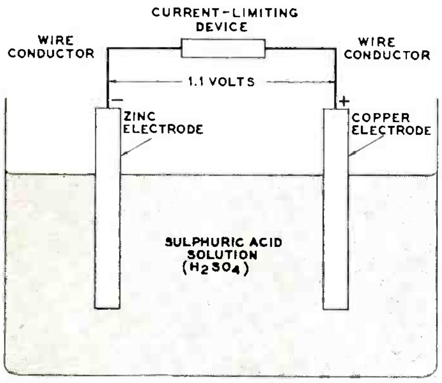


Fig. 2. Cell similar to that in Fig. 1, but single solution causes polarization.

acid solution, for which the ions are the negative sulphate radical and positive hydronium ion (a combination of a water molecule with the hydrogen ion). Operation is similar to that in a displacement cell. However, not copper but hydrogen is formed at the copper electrode. The layer of hydrogen effectively alters the material of the electrode, and a reverse emf is set up which reduces the output voltage of the cell. Polarization is the name applied to this process: it is usually counteracted by the addition of a substance (a depolarizer) that combines chemically with the hydrogen as it forms. A more practical version is the commercial dry cell, shown in Fig. 3. The negative electrode is zinc, which is used as the container of the cell; the positive electrode is a carbon rod in the center, which is insulated from the can. A lining of absorbent paper is placed within the can, which is then filled with a mixture of ammonium chloride, zinc chloride, powered carbon, and manganese dioxide. The battery is sealed with pitch or a similar airtight and moistureproof substance. Often a substance is used to turn this mixture into a jelly-like paste; hence the term "dry cell". The zinc, as in other cells, goes into solution as ions, leaving electrons behind. Ammonium and hydronium ions, migrating toward the carbon electrode, receive electrons and would become ammonia and hydrogen gas if it were not for the depolarizer, manganese dioxide. This combines with them to form ammonium hydroxide and manganous oxide. The potential difference across the terminals of a dry cell is about 1.5 volts, and falls with continued use.

Before leaving the subject of electric cells and batteries, a few other points should be made. First, the discussion has been confined to non-reversible batteries; that is, cells that cannot easily be recharged. Storage batteries — the kind used in your automobile — operate on the same general principle but can be recharged by forcing current through them in the direction opposite to that with which they discharge. However, they are bulky, heavy, and use a corrosive liquid electrolyte, and so are not ordinarily employed in electronics. Second, batteries provide single-direction current (DC) only. Third, the voltage or potential difference produced by a cell is dependent only on the materials from

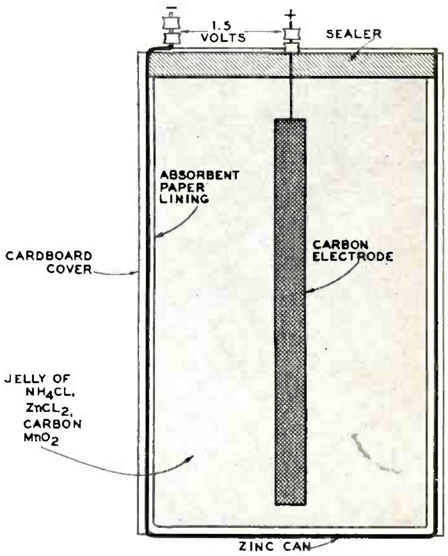


Fig. 3. Practical cell: a dry cell battery.

which it is made — not on its size. Larger cells are, of course, usually able to furnish more current or the same current for a longer time than smaller cells. And individual cells may be combined in one package and connected so that the emf's are additive, if a larger emf is desired; or they may be combined so as to have the same voltage as one of the units but more current capacity.

SPEAKER ENCLOSURES

Continued from page 20

peaks will appear at *two* frequencies. The size of the port is then adjusted until repeat tests show the 2 peaks to be of approximately equal voltage. When this happens the resonance of the speaker and the resonance of the enclosure are matched.

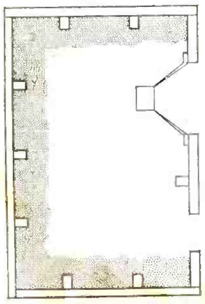


Fig. 8. Bass-reflex cabinet is smaller than usual total enclosure, and has an opening or port.

The port adjustment may be made in a variety of mechanical ways. In my own experience I have found one method to be superior: cover the port with a board whose entire surface has been drilled with ½-inch or preferably ¼-inch holes, and then fill in as many holes as called for, using plastic wood or some similar substance. The advantage of this method lies not so much in its convenience as in the fact that the holes constitute at least part of the acoustical damping that is required.

After the voltage peaks have been made equal, layers of loosely woven cloth such as burlap should be stretched across the port for damping, and the tests made again. A combination of the proper number of layers of the proper cloth will reduce the height of the peaks until the voltage reading remains almost uniform over the entire run. It is sometimes extremely difficult to achieve this condition with a simple port, where the interstices of the cloth must provide all the damping, but it is usually relatively easy with a port consisting of a large number of holes or slits.

Another method for tuning a bass-reflex enclosure (using the same procedures for varying the port area and damping) does not require a signal generator and voltmeter, but does require an oscilloscope and a dry cell, which can be a standard flashlight battery. The scope is connected directly across the speaker terminals, as in Fig. 10A; the speaker is disconnected from the amplifier and stimulated by intermittently touching the wires to the terminals of the dry cell.

With a properly tuned and damped system a pattern like that of Fig. 10B will appear momentarily on the screen at the circuit "break", showing that the speaker voice-coil returned to its normal position with very little ringing or hangover after the stimulating voltage was removed. The screen pattern is produced by the counter-emf of the loudspeaker voice-coil and magnet, which act as a generator.

With an improperly tuned and damped system, however, the screen pattern looks like that of Fig. 10C. It is a little more difficult to make all adjustments by this method, as the effects of tuning and damping are not shown independently. The method does, however, constitute an excellent final check. Furthermore, those with mechanical ingenuity can convert an ordinary 6-volt house doorbell into a low-speed circuit breaker, working at a frequency of 5 cycles or so (the bell must be removed and the clapper heavily weighted). The oscilloscope sync controls can then be adjusted for a permanent damping pattern on the screen, and port adjustments can be made while actually watching their immediate effect on the screen pattern.

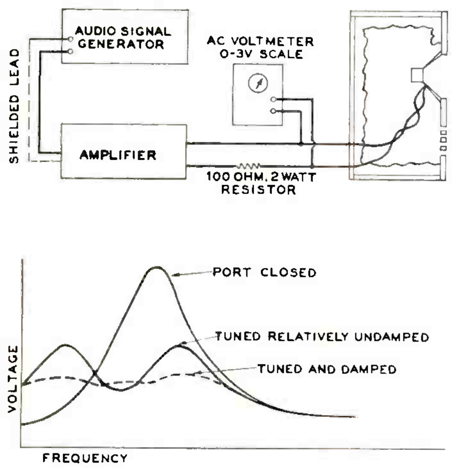


Fig. 9. Usual setup for tuning bass-reflex.

The Horn

Acoustical horns constitute a subject requiring more detailed treatment than can be afforded here, so the general principles involved will be mentioned only briefly.

Basically a horn is a tube of progressively increasing diameter. When a sound source is placed at the narrow end, or throat, it engages *all* the air of the horn, and the effective radiating area of the source may be increased approximately to that of the horn's wide end, or mouth. This has the effect of greatly increasing the efficiency of radiation compared to direct-radiators, and the same sound energy can be radiated with much smaller cone or diaphragm movements.

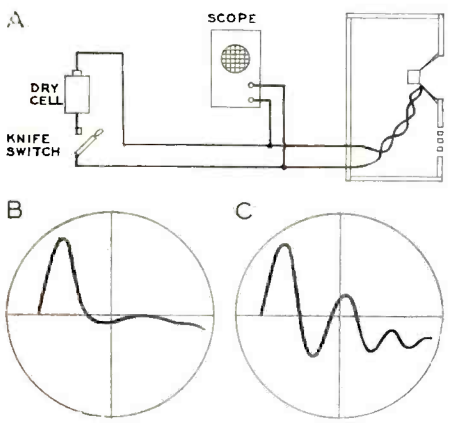


Fig. 10. An alternative setup using 'scope.

Horns have 2 characteristics which are fundamental to their use with loudspeakers. One is the bass cutoff frequency, below which response falls off rapidly. This is determined by how quickly the horn diameter increases; that is, the rate of flare. The other characteristic is the mouth area — the smaller the mouth (in relation to a particular frequency) the greater the sound reflections and air-column resonances set up, with their attendent peaked response and boominess.

A good low-frequency horn must have a slow rate of flare and a large mouth area. This adds up to a *very* large structure. By using the corner of a room, and allowing the walls to continue the horn flare (mitigating the abruptness of the transition from the horn mouth to the room), these dimensions have been reduced to practical figures.

LOAD IT WITH SAND

Continued on page 32

that it might be suitable for speaker panel damping. It has the advantage of weighing roughly less than a fifth as much as sand by bulk, but it is not so easily obtained and is much more expensive. Moreover, in simple comparative tests it proved to be about a fourth as effective in damping properties as sand. It might be fully as effective if the same weight were used, but this would involve too much space and expense.

Sheet lead bolted between 2 sheets of plywood has been tried, but the principle gain was in weight, without greatly improving damping. Concrete or brick construction is of course quite vibration-proof, but useful only in built-in installations. I have concluded that sand loading is probably the greatest single improvement which can be made to the average sound installation, taking into consideration the small amount of money and labor which it requires.

I have an acquaintance who owns a low-priced corner cabinet, in which he uses a 12-inch speaker of rather indifferent quality. He had tried all kinds of bracing, even 2-by-4's, trying to prevent baffle board vibration. I recommended sand loading. By simply adding a new front of ½-inch matching plywood set out on one-inch wood strips, with a square frame around the speaker hole, and packing the space with sand, he achieved a remarkable transformation. The bass was much improved, and the whole spectrum made smoother and more pleasant to listen to.

In my own installation I use a separate bass enclosure of about 20 cubic feet, mounting two 15-inch woofers. I had tried internal bracing and lining the whole enclosure with an extra layer of ¾-inch mahogany, plus a lot of insulation, but never quite got rid of some annoying vibration. However, the addition of a box inside the speaker panel at each end, both boxes filled with sand to a thickness of 1½ in., really did the trick. Of course, no one has tried to move the enclosure since.

The more powerful the bass, the more you may need sand loading, for it is the slow, heavy vibrations which benefit most from panel damping. There is no reason why sand loading should not be equally helpful to large exponential horns, and it should be just the thing for the mighty air coupler.

If it is a built-in job which will never have to be moved, and you are very ambitious, you can fill the remaining panels with a thin layer of sand, as there may still be some minor vibrations in them after loading the speaker board. However, most of the disastrous vibration in an enclosure would seem to originate in the speaker panel through the direct push and pull of the speaker mounting. A shaking speaker panel transmits its movements to the rest of the enclosure, while comparatively little disturbance is caused by the compression and decompression of the air within the chamber; damping the remaining panels, then, is of minor importance.

Construction of a sand loaded speaker panel, Fig. 1, is very simple. For 15-inch woofers, and especially for multiple-woofer enclosures, use ¾-inch plywood to mount the speaker. For smaller speakers, ½-inch material is good enough. After cutting the speaker hole, screw and glue strips of ¾-inch wood on the back of the board to form the hollow interior. The thickness of the space thus formed should be from ¾ to 1¼ in., depending on the power and number of speakers used, though 1 in. is usually sufficient to deaden any vibration for a single-speaker panel.

First fasten the strip across the base of the panel; then run strips up the outer edges. Two more vertical strips are fixed on either side of the speaker hole, far enough apart to allow the speaker to be mounted comfortably. Make these strips ¾ in. short at the top, to allow for the closing strip to go in. Two horizontal strips between them now form the square around the speaker hole. (If you have a bass reflex port in your front panel, it must be similarly framed.) Fasten the top one, but leave the lower one out, so that the bottom section can be filled. It is well to fasten a small stop block on either side of this section, so that the closing strip cannot go in too far.

Now screw and glue ½-inch plywood over the panel, leaving the square window around the speaker hole uncovered. Stand the panel upright and with a funnel pour in the sand, filling all 4 sections, and tamp well.

The sand should be of the grade used for plastering — clean, free of gravel and dust, and *dry*. If it is not dry enough to run freely, spread it out thin on a clean concrete driveway in hot sunshine, until it is. Lacking a concrete driveway and hot sun, put the sand in shallow roasting pans and bake it in the

Continued on next page

LOAD IT WITH SAND

Continued from preceding page

kitchen range. If your wife has any objections to this plan, read the foregoing to her, and she will of course understand at once the importance of this project, and co-operate. It takes only a few pans full for the average baffle.

After filling with sand, drive the top closing strips down to the stop blocks and fasten with glue and screws from both sides. Inspect for any tiny cracks or openings which might allow sand to escape; if any are found, seal them with plastic wood.

Now if your enclosure is otherwise properly designed and constructed, you are about ready to enjoy the full range and smoothness which the manufacturer has built into your speaker. You will also probably acquire an irritating habit of rapping your knuckles on the speaker panels of your friends' baffles, and smugly comparing the sharp, hollow, ringing sound thus produced with the fine, satisfactory, dull thud that you hear when you pound on your own.

MICROPHONE MIXER

Continued from page 27

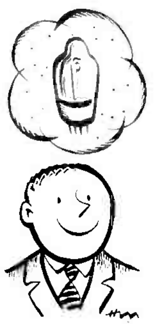
phasing between them. Then after the mixer is completed and found to operate properly, connect three input plugs in parallel, making certain that each lead goes to the same pin number on each plug, and insert them all into the input receptacles. Then feed a steady tone into all three inputs simultaneously, using a signal generator or test record, and turn one of the mixer controls to its half-way setting, leaving the others fully off. Set the recorder's volume indicator for about a half-scale reading, and slowly bring one of the other mixer controls up to its half-way mark, noting any change in the record volume indicator reading. If the reading increases, those two channels are in phase with each other. If it decreases, they are out of phase, and the primary connections to one of them should be reversed. When these are correctly phased, turn one control off and check out the third channel in the same manner.

Then the phasing of the microphones can be checked, a pair at a time, by placing them close together and feeding them a constant tone from a loudspeaker. Repeat the previous adjustment procedure, with the mikes replacing the paralleled input plugs.

Apart from the phasing check, the only thing that may require adjustment when the mixer is completed is the hum-balancing potentiometer on the power supply. This should be set with the mikes plugged into the mixer and the mixer controls turned up high enough for the hum to be audible through the recorder's monitor headphones. Any attempt to make this adjustment using a monitor loudspeaker will result in gross feedback. If the phones will not respond to the hum signal it may be possible to use the recording indicator, setting the hum control for minimum meter indication. If the ambient noise level in the room is found to be too high, so that it overrides what hum might be there, it is suggested that the mikes be put under a couple of thick pillows or folded blankets on a couch.

Conclusion

It is rather difficult to evaluate the subjective advantages of a low-distortion mixer until some of the tapes made from it are compared with those made on less ambitious equipment. The difference may best be described as an increase in "transparency" and definition, and a noticeably smoother effect from highly transient sounds. Massed strings take on a silky quality that is rarely heard from records and is not too common from live tapes either. And the "edgy" sound that accompanies a singer's normal tones (the so-called vocal production) is significantly less annoying, permitting more intimate mike placement if desired.



One of the more startling effects noticed with a truly fine input transformer is the apparent smoothing out of microphone peaks. This, together with the low distortion in the following stages, probably accounts for the sheen of the resulting sound.

PARTS LIST

Resistors:

1	4.7 Mohms	½ watt carbon
1	2.0 M	½ watt carbon
1	1.0 M	½ watt carbon
3	470,000	1 watt low-noise
1	270,000	½ watt carbon
3	100,000	1 watt low-noise
1	47,000	½ watt carbon
1	39,000	2 watt carbon
1	33,000	1 watt carbon
1	30,000	5 watt wire-wound
1	6,800	1 watt carbon
1	2,500	5 watt ± 5% wire-wound
3	500	1 watt carbon
3	500	1 watt wire-wound (IRC BW-1 or equivalent)
3	100,000	audio taper potentiometers
1	250	5 watt linear wire-wound potentiometer

Capacitors:

4	.05 mfd 600 volt tubular paper
4	.25 mfd 600 volt tubular paper
1	20, 20, 20 mfd/450, 450, 450 volt can electrolytic.
1	20, 50 mfd/450, 250 volt can electrolytic.

Plugs, Sockets, Connectors:

3	Cannon XL-3-13 chassis mount receptacles (J-1, J-2, J-3)
1	Cannon XL-3-14 chassis mount receptacle (J-4)
1	Cinch-Jones P-306-AB chassis-mount plug (P-1)
1	Cinch-Jones S-306-CCT
4	Noval tube sockets
1	7-pin miniature tube socket
1	AC power plug

Chokes, Transformers:

3	Peerless K-241-D input transformers (T-1, T-2, T-3)
1	Merit P-3047 power transformer or equivalent (T-4)
1	8.5 henry 400-ohm 50 milliamp choke (L-1)

Fuses, Lamps, Switches:

1	1 amp. 3AG fuse
1	Fuse holder for chassis mounting.
1	Dialco Series 710, No. 431 pilot lamp mounting or equivalent.
1	No. 47 pilot lamp
1	SPST bat handle toggle switch

Tubes:

3-12AY7, 2-12AX7, 6X4

Miscellaneous:

3 Dial plates calibrated 0–10 or 0–100 for 270 degrees rotation.
Terminal strips as required.
Aluminum chassis 13 by 5 by 3 in.
Aluminum sheet heavy gauge, 16 by 6½ in.
Mini-box, 5¼ by 2 by 3 in.
½-inch 3-ply wood sheet, 4 by 4 ft.
Carrying handle
8 Suitcase clasps
Bolts, nuts, washers, solder lugs, rubber grommets, sheet metal screws as required.
3 Knobs
8 Rubber ft.

AUDIOCRAFT Test Results

On instruments, the mixer tests out very well although its distortion is not quite down to the level of the finest existing playback equipment. Its 0.6% IM distortion at 1 volt out is, however, considerably lower than that from most of the costliest units commercially available. Other performance results:

IM distortion (60 and 7,000 cycles, 4:1): 0.3% at 0.25 volt output; 0.6% at 1 volt output; 1% at 3 volts output.

Response: ±1 db, 15 to better than 25,000 cycles.

Input level for 1% IM: —36 dbm.

Gain: 57 db.

Output impedance: 600 ohms effective, when terminated by at least 100,000 ohms.

Used with properly phased microphones, the mixer can be operated as a fader also; there is no interaction among the individual volume controls, and no quality deterioration at intermediate settings.

To maintain a healthy safety margin for all the equipment concerned, it is recommended that the half-way rotation point be used as the normal full-on setting for the mixer. The channel that has the highest level fed into it from a microphone should, then, have its control set at the half-way mark, and the other controls may be set to vary the desired balance around this point. The recorder's volume control can be used as the master gain control. Adoption of the mid setting as the "normal" mixer position is a precaution against overload of either the mixer stages or the recorder's input stage, and guarantees that the distortion from either source will never get above its minimum level.

Total cost for parts: about $185.00.

TV CAN SOUND BETTER

Continued from page 17

quite adequate. It takes 2.7 volts to drive the unit to full output (4 watts). Hum and hiss are completely inaudible 6 in. from the speaker.

Such performance is particularly noteworthy in view of the fact that all parts were "standard" — no specially selected or matched components were used. It

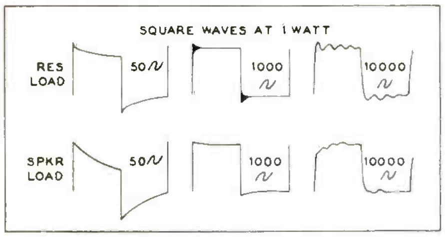


Fig. 7. TV amplifier square-wave results.

can be confidently assumed that equivalent performance would be obtained by anyone buying and assembling the parts in the usual way. There is no reason why the amplifier should be limited to use with TV sets, needless to say; standard radio or phonograph consoles would benefit also by modernization of this type. And, with a small power supply, this would make an excellent medium-cost amplifier for a modest hi-fi installation.

CHASSIS LAYOUT

Continued from page 23

hooked up. Might be a good idea to use a cheap speaker for these first tests, just in case the leads of the output transformer were accidentally reversed and you get positive feedback instead of negative. Also, set all the controls to zero before turning on the power.

This is the moment you've been waiting for (in fact, I'm sometimes so eager that I haven't even finished wiring the gadget) and, if everything has gone well, satisfying sounds will come from the loudspeaker. Unfortunately, there are also a few other possibilities: nothing lights up, because you forgot to put a fuse in the holder or made a mistake in wiring; perhaps the tubes light up, but nothing else happens; maybe you get weak, faintly recognizable sounds from the speaker, or possibly a tremendous roar or siren-like scream because of positive feedback.

If the end results don't quite come up to your expectations, you'll need to do some trouble shooting, a subject that we'll deal with in detail in a subsequent issue. If you don't care to wait that long, look up an electrically inclined friend and get him to help you out while you watch over his shoulder. Briefly, you should check first for mistakes in wiring, then for faulty components. Then too, perhaps the equipment that you're using to test the amplifier isn't connected properly.

In any event, don't give up. Building your own equipment can be a source of real satisfaction and enjoyment. The first few chassis may look ungainly and have too many holes, the wiring may look like an impenetrable maze, and the perfomance may not be the greatest. But keep on, and as you gain confidence, skill, and familiarity with equipment, you'll start to turn out work that you'll be proud of.

GROUNDED EAR

Continued from page 5

ful application of the D'Arsonval movement in the Electro-Sonic cartridge. Now Angel is producing a pickup which, though labeled moving-coil, is actually a moving-needle (or moving-armature) device. The moving-armature principle in the past was used in the most sensitive galvanometers because it possessed the best possible response by virtue of requiring the least mass in the movement. The same virtue offers great promise in pickups.

In the moving-coil pickup, the needle has to *carry* a coil and move this coil within the poles of a magnet, thus cutting the magnetic flux and generating a current in the coil proportional to the motion. In the moving-armature movement, on the other hand, the coil is stationary. It surrounds the needle (or rather a small vane or flag attached to the needle). The needle alone moves and its movement vibrates the flag or vane — the armature — which generates a current in the coil proportional to the velocity. The absence of a coil on the needle reduces the required mass of the needle and its support, and compliance can be increased proportionately. The theoretical compliance is nearly unlimited so far as phono pickup use is concerned. Moreover, since the coil no longer needs to be small, it can be designed for a size dictated by other considerations: efficiency, sensitivity, capacitance, and so on.

The Angel is, therefore, interesting not only for its performance (which, incidentally, is excellent indeed), but because it introduces another type of movement for future development and refinement.

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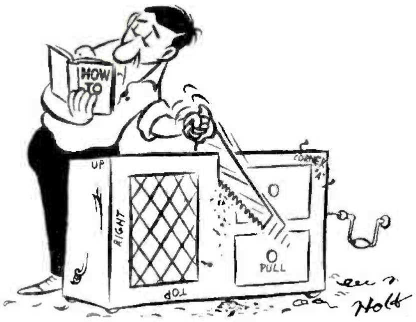

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WOODCRAFTER

Continued from page 8

1) The hammer handle is long, and purposely so — don't grasp it halfway; hold it firmly at the end.

2) Make the first swing of the hammer a light one to set your aim. Subsequent blows should have more impact.

3) Hit the nail head squarely to prevent bending the nail or glancing off and denting the wood. The hammer handle should be at 90° to the nail at the moment of impact.

4) When driving small nails or brads, use only a wrist movement. When driving larger nails, use the elbow and shoulder as well.

5) Drive the first nail straight and the remainder at an angle. The first nail, driven straight, will bind the 2 pieces of wood without slipping them out of position. The remaining nails, driven at an angle to the surface, will have more holding power than if driven straight.

6) When nailing in line with the grain of the wood, stagger the nails to avoid weakening and splitting the stock. This is especially true of plywood.

7) If a nail should bend, drive in another nail about an inch away before pulling out the bent nail. The bending of the nail might be caused by a knot or poor grain. If the second nail also bends, remove it and drill a pilot hole smaller than the nail. Then you will be able to drive a new nail straight. Whatever you do, don't try to hammer a bent nail.

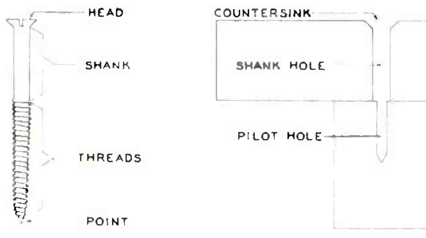
8) Cabinet work requires finishing nails with a slender head that can be set below the surface of the wood. With the hammer drive the nail until it is about 1/16 in. above the surface. If you hammer it beyond, you run the risk of marring the wood. Then with a nail set drive the nail about 1/16 in. below the surface, the remaining hole being filled later with wood putty or plastic wood.

9) In pulling a nail slide the hammer claws snugly against the nail and, with a small block of wood under the head of the hammer, apply firm leverage until the nail is removed. The block of wood adds to the leverage and prevents marring the stock.

10) The face of the hammer should always be kept clean to prevent its glancing off the nail head. Glue or other foreign matter can be removed by rubbing the hammer face against a brick or sandpaper.

How did you score on the hammer test? Before you've had a chance to grade yourself here, we'll jump right into the subject of driving screws. This is a more complicated problem than it might appear on the surface, even though driving screws into soft wood is very simple. Just a gentle tap on the head of the screw to start it on its way, and the turn of the screwdriver finishes driving it home. It's a bit different, however, when working with hardwoods and plywood. Here it is necessary to remove that part of the wood which will be occupied by the screw. An attempt to drive the screw into hardwood without first drilling to make room for it usually ends in disaster, with the wood splitting and the finish marred or the screw itself broken or damaged.

To assure a good job 3 operations are necessary. First, a hole should be drilled in the wood to receive the shank



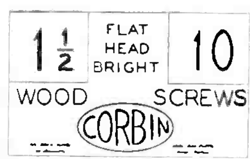
Wood screw and the proper way to use it.

of the screw, the unthreaded part. Second, the top of the hole should be made into a larger tapered opening by using a countersink so that the head of the screw will be flush with the surface of the wood when it is driven home. The third and last operation is to drill a small pilot hole through the area where the threaded portion of the screw will be driven. This must necessarily be somewhat smaller than the diameter of the threads or the screw won't grip.

To determine the sizes of drills needed for shank holes and pilot holes for the most commonly used screws, the following table will serve as a guide.

Screw Gauge No.	Shank Hole	Pilot Hole
2	3/32 in.	1/16 in.
3	1/8	1/16
4	1/8	1/16
5	1/8	3/32
6	5/32	3/32
7	5/32	1/8
8	3/16	1/8
9	3/16	1/8
10	3/16	1/8
11	7/32	5/32
12	7/32	5/32

In your hardware store the label on the end of the box of wood screws looks like this:



The number at the left ($1\frac{1}{2}$) indicates the length of the screw in inches. The number at the right (10) indicates the thickness of the shank. Thus, the table shows that a number 10 gauge screw needs a 3/16-inch shank hole and a pilot hole of ⅛ in.

Before driving a screw dip the thread into wax or oil and you will find the turning much easier on both your muscles and the head of the screw. Use a screwdriver that fits the slot in the

Continued on next page

WOODCRAFTER

Continued from preceding page

head of the screw: one that is either too big or too small will mar the head or slip off and damage the wood.

Planes and chisels have much in common and their use cannot be covered briefly. In the near future we'll give them the time and consideration due such interesting and valuable tools. Which reminds me that tools are extremely welcome Christmas gifts — they make a man handy and happy. May your holiday season be the nicest yet.

SOUND SERVICING

Continued from page 10

wire system (2 inside wires, one shield), as follows:

a Former inside wire (the grid or "hot" wire) to one of the 2 inside wires.

b The other side of the cartridge to be "lifted" (i. e., disconnected) from the changer, arm, or chassis ground, and then to be run as the other inside wire straight to the phono plug. This is then the AC ground.

c Turntable chassis, arm, and cartridge case to be connected by a third wire to the cable shield.

d At phono plug, wire *a* is connected to the inside contact; wire *b* and the shield (*c*) are twisted and connected to the shield of the plug.

This method of wiring will often help in cracking hum caused by ground loops. or from varying AC potentials between chassis. It may also help to eliminate RF-induced hum.

7) If you live in or near a large city, with several TV and FM transmitters, or simply happen to live very near one of them, you may find that your phono preamplifier is perpetually tuned in to one transmitter or more — you get the program along with the music you are trying to play. In other cases, even though the stations themselves aren't audible, stray RF energy may cause annoying and very hard-to-diagnose noises, either as hum, buzzes, or combinations of such disturbances. The following approaches have all been used, with varying success, on this problem that is little recognized though fairly common.

a First solution is to have your phono preamp checked. A gassy tube, or one running with wrong bias values, will tend to act as a detector and pick up these signals. Correcting the bias condition will also reduce distortion.

b The 3-wire system outlined above may help. Sometimes, it has been found, the third (shield) wire can be cut loose from the phono plug and grounded to the preamplifier chassis at random points: one of them may suddenly wipe out the noise.

c Try random reorientation of all your input and intra-chassis wiring, even those from tuners, tape recorders, and such. In one case, for instance, it was found that RF pickup was reduced by changing the position of wires from the preamplifier power supply into the preamplifier! Probably the reason for these differences is that certain cables tend to act like loop antennas: breaking up the loops, or changing their directivity, will reduce the RF pickup effect.

d Change to a cartridge of different manufacture. The higher the output, the less the RF problem, in general — that old business of signal-to-noise ratio. But in other cases changing to a cartridge

PROFESSIONAL DIRECTORY

TRADER'S MARKETPLACE

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

RAM CORNER ENCLOSURE, cabinet unfinished containing 5- and 8-inch Wharfdales and 12-inch Bozak speakers and complete crossover network; $150.00. Waveforms A-20-5; Amplifier and separate Preamplifier. Used one year. New $199.95. Now $50.00. Both $190.00. FOB, Great Barrington. W. B. Syer, The Publishing House, Great Barrington, Mass.

with lower output reduced the problem; the reason, probably, is that the new cartridge had a different inductance, which changed the frequency of tuning.

e Install an RF filter in the phono input line, a low-pass filter rolling off sharply all response in the RF region. Several preamplifier manufacturers, particularly those selling in the congested New York area, have already incorporated them in production units.

8) Other kinds of hum-bucking techniques. Every so often there occurs a kind of hum problem that seems to be caused by differences in the levels above ground of the various interconnected units. For instance, you probably know that often one position of the record player's AC plug, in relation to the preamplifier's AC plug, will produce less hum. Other examples of this kind of phenomenon are:

a Phono hum lessens when tuner's AC plug is reversed (or that of other high-level input devices).

b Phono hum lessens when tuner's audio input plug is plugged into preamplifier — or the hum in certain cases increases.

c Phono hum diminishes when the tuner is left on.

If you have ever by accident touched the chassis of your preamplifier, and then that of your tuner, both on and with no ground connected between them, you have probably experienced directly the causes of these obscure hum differences — the differences in potential between chassis. The technique of straightening out these sources of extra hum is, frankly, intuitive. Sometimes grounding the amplifier chassis to the power line through a 0.1 μfd condenser will reduce the hum; sometimes removing such a ground from one piece of equipment will reduce the hum! Some of these noises seem more prevalent when the AC wiring is old and decayed, perhaps because the house ground has become defective.

Next month will be discussed the kinds of noise that develop within the equipment, and suggested cures.

EDITORIAL

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feature 1-mil or ½-mil microgroove styli, track at 2½ to 5 grams.

Rek-O-Kut Company — An arm so new that no literature was available, but which seemed capable of exceptionally fine performance.

H. H. Scott, Inc. — The 710-A stroboscopic turntable, seen under glass last year, is now in full production.

Shure Brothers, Inc. — Series ML44 (Music Lovers) dual-play ceramic pickup cartridges can be used to feed high-level input channel directly or, with adapter supplied, can be connected to magnetic phono input.

Sonotone Corp. — Latest turnover ceramic cartridge from Sonotone is the 3 series, with increased compliance and reduced tracking force.

United Audio Products — The Dual Model 1003 record changer appears to have all the automatic convenience features ever conceived, and to do a good job of playing records as well.

Weathers Industries, Inc. — A new viscous-damped arm, a redesigned cartridge and stylus assembly that appears substantially more rugged, and a modulator system (inductively tuned) that is claimed to be completely stable.

Speakers, Speaker Systems

Acoustic Research, Inc. — Now in full production on acoustic-suspension woofers and complete systems.

Altec Lansing Corp. — Two new Iconic speaker systems, one using a 15-inch woofer and the other a 12-inch woofer. Both are bass-reflex systems. Also a redesigned version of the deluxe 820 system series, the 820C.

Beam Instruments Corp. — New 10- and 12-inch Duplex coaxial speakers with mid-range stabilizers which consist of oval thickening patches on cones. Also a line of corner and wall-mounting bass-reflex enclosures for speakers of from 10- to 15-inch diameters.

British Industries, Inc. — A new Wharfedale tweeter, only 3 in. in diameter.

Brociner Electronics Corp. — Transcendent and Model 4 systems employing new Lowther PM-3 twin-cone driver; Lowther TP-1 complete corner horn system.

Electro-Voice, Inc. — E-V horn-loading speaker systems added this year are the Centurion, a complete 4-way system, and the Empire, available in 2-way and 3-way systems. Plans or complete kits are now available for several E-V systems.

General Electric — Added this year to the GE speaker line are the 850, an extended-range 8-inch speaker, and a bookshelf enclosure for it.

International Audio Group — Several combinations of Bozak speakers in custom-built enclosures.

International Electronics Corp. — Completely new line of Frazier-May speaker systems utilizing radial slot dispersion of high frequencies. All are priced under $100.

Janszen Mfg. Co. — Push-pull electrostatic tweeters for crossover at 500 or 1,000 cycles, for external or built-in applications. Also complete systems using the electrostatic tweeter and 2 Bozak woofers, specially baffled.

Jensen Mfg. Co. — The PR-100 Imperial and RS-100 Laboratory Standard deluxe speaker systems; TV Duettes, for use with television; a complete line of kits for these and other Jensen systems.

Kingdom Products, Ltd. — New Lorenz LP-312 12-inch woofer; TB-1 and TB-2, 1 and 2 Lorenz tweeters on coaxial-mounting brackets; LP-312-1 and LP-312-2, coaxial and diaxial combinations of 312 with TB-1 and TB-2, respectively.

Klipsch and Associates — The Klipsch Shorthorn is now available as a finished 3-way system with 12- or 15-inch bass driver, or as an enclosure only.

James B. Lansing Sound, Inc. — Hartsfield deluxe speaker system; several extended-range loudspeakers.

R. I. Mendels, Inc. — Extended-range 8-inch loudspeaker, the Panasonic, imported from Japan.

John Ould (U. S. A.) Ltd. — Pamphonic Senior and Pamphonic Junior deluxe loudspeaker systems employing wall dispersion of high frequencies.

Permoflux Corp. — New this year is the Largo 12 system, employing 12- and 8-inch speakers and the 32KTR tweeter. Also demonstrated was the Permoflux Stereovox

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ADVERTISING INDEX

EDITORIAL

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technique, which achieves liveness by artificial reverberation.

Pickering and Company, Inc. — Two deluxe electrostatic speakers; one model for 1,000-cps crossover and the other for 400 cps. Both have continuously-adjustable level controls.

Plastilex Products, Inc. — The Bonn Sonosphere is a small spherical speaker enclosure that can be mounted anywhere. Enclosure is of flexible plastic, so that entire surface radiates at bass frequencies.

Racon Electric Company, Inc. — A series of new 15-inch loudspeakers: Model 15-HW woofer, 15-HD dual-cone extended-range, and 15-HTX triaxial. All employ unique edge-suspension and plastic cone-stiffeners to achieve low resonance with high efficiency.

Sherwood Electronic Laboratories — The Forester 3-way speaker system, employing cone-type mid-range speaker and tweeter, with horn-loaded woofer.

Stephens Mfg. Corp. — The tweeter previously used only in the 152AX and 122AX coaxial speakers is now available separately as the model 212.

United Audio Products — A line of low-resonance woofers, tweeters, and extended-range speakers. Made by Wigo, the speakers feature impregnated glass fiber construction at critical points.

United Speaker Systems — The Premiere speaker system, with completely horn-loaded theater-type driver components.

University Loudspeakers — A greatly expanded line of coaxial and triaxial speakers, tweeters, and universal divider networks. Also the Dean and Classic deluxe 3-way systems, and a line of horn-loaded phase inverter enclosures.

Weathers Industries, Inc. — Model SE-100 speaker system, a combination of 12-inch woofer and 3-inch tweeter in a remarkably shallow enclosure designed for wall mounting.

Tape Recorders

Ampex Corp. — The Ampex Model 612 monaural/binaural tape playback unit uses stacked binaural heads. With 620-type amplifier-speakers, it makes a compact high-quality system.

Ampro Corp. — New Career and Hi-Fi 2-speed recorders, and the Hi-Fi Consolette with 2-way speaker system and tape storage space. Console speaker systems are available, and built-in AM receivers are optional.

Bell Sound Systems, Inc. — Model RT-88 is a 2-speed, 3-motor recorder with piano-key controls. Weighs only 27 lbs.

Berlant-Concertone — New are the BRX and BAX series, similar to the BR series, but revised and improved. The BAX has provisions for local or remote-control operation.

Daystrom Electric Corp. — Model 360 Crestwood recorder can be used as a table model or, with legs, as armchair consolette. Features are a 10-watt power amplifier and a built-in 2-way speaker system.

Fenton Company — Fen-Tone MoTek tape transport mechanisms are imported from England; there are tape preamplifiers to match. These are inexpensive 3-motor tape decks.

Pentron Corp. — New Pentron recorders are available from the low-priced Clipper, through the medium-priced Pacemaker, the deluxe Emperor, and the professional Dynacord. The Emperor, incidentally, has a 10-watt push-pull amplifier.

Presto Recording Corp. — New is the SR-27 tape recorder, for professional or home hi-fi applications. Consists of the R-27 3-head tape transport mechanism and the A-920B amplifier in separate cases. Also a line of redesigned high-quality turntables.

Revere Camera Company — Among new models is the medium-priced T-1100, with a built-in radio at extra cost; an automatic slide projector and a projector-recorder synchronizing device; the T-700-D deluxe portable recorder; and the T-11 professional chassis for home hi-fi applications, which will handle 10½-inch NARTB tape reels.

Telectrosonic Corp. — The Telectro Custom 220 tape deck looks interesting at less than $100. At the other price extreme is the new Professional 1000 model, a broadcast recorder. Telectro has also a binaural/monaural tape player and a pair of bookshelf speaker systems to go with it.

Viking of Minneapolis — Using Dynamu heads, the Viking 75 tape deck and preamplifiers have impressive specifications and very low prices. — R. A.

TAPE NEWS

Continued from page 14

on his machine, is a measured flat frequency response all the way to the machine's normal limit. The assumption then is usually that the head is correctly aligned.

Of course, when he passes one of his tapes on to someone else, whose machine happens to have the same equalization characteristics and a *correctly* aligned playback head, the sound is gloriously mellow, due to negligible response above 3,000 cycles.

The only way to check head alignment or azimuth adjustment accurately is with a commercially-produced alignment tape.* These tapes are recorded on a machine that is known to be in perfect alignment, and they carry a sustained high-frequency signal that is attenuated by any mis-alignment of the playback head. To align the head, play the tape and turn the head's adjusting screw one way or the other until maximum output is obtained from the alignment tone. The final adjustment of the screw should always be clockwise, so that there will be no tendency for the head to get "hung up" on the screw threads and then slip back out of adjustment when the unit is jarred some time later. A few of the earlier model recorders have no provision for head alignment, but for others there is usually a note in the service manual as to the procedure to follow with that particular recorder. Current production models are almost invariably aligned before delivery, and these will be found to have small gobs of cement holding the adjusting screws. Where the screws are so fixed, there is no need to check the alignment unless you strongly suspect that something is out of kilter.

Alignment tapes are slightly expensive, as recorded tapes go, so if the absolute maximum high-frequency response is not deemed worth the expense, you can do fairly well just by playing a pre-recorded music tape and aligning the head by ear, setting for maximum audible high-frequency response. For this test, do *not* try boosting the treble to make it "more readily audible." The ear is more sensitive to increased frequency range when the over-all response is as close to flat as possible.

*Such as Ampex's No. 5563 for 7½ ips tapes.

Picking Winners...

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Superlative 3-way system, deluxe 602

Employs two fine Stephens 103LX 15" low frequency drivers featuring unusually large spider assemblies and 4¼ lb. Alnico V magnets. Molded straight sided cones and 2" voice coils afford faithful bass response down to 20 cps. System includes P-30 high frequency driver, delivering full 30 watts above 600 cycles; 214 ultra high frequency driver with precision handspun dural diaphragm for 5,000 to 22,000 cps range; 625H true multicellular midrange horn. Crossovers 600X and 5000X with attenuator. Complete system. **Net $393.75.**

Tops in 2-way systems, super 803

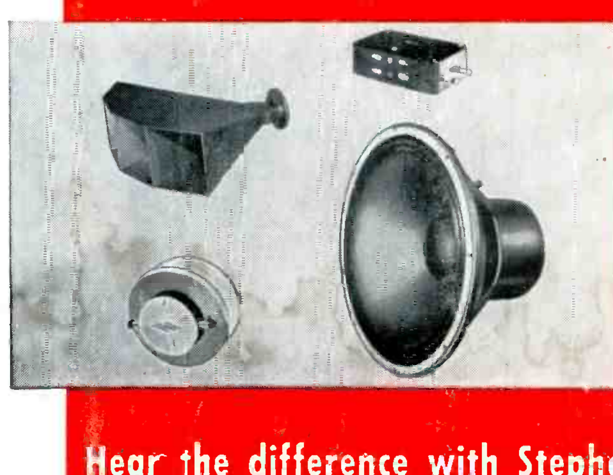
Uses two 103LX low frequency drivers, the finest available (and same as in 3-way system above), a Stephens 216 high frequency driver releasing full 20 watts above 800 cps. Horn is the multicellular 824H, 2 cells high and 4 cells wide. System 803 utilizes 800X crossover and attenuator. Frequency range extends from 20 to 18,000 cps. 30 watts power capacity. **Net $269.25.**

Note: This speaker system converts to a three way system with the addition of a Stephens 214 super tweeter and 5000X network.

Best for the money, 2-way system 801

Low frequency driver is 15" 105LX, with a 2½ lb. Alnico V magnet, 2" voice coil, large spider assembly and sturdy cast aluminum frame. System 801 has a 216 high frequency driver, 814H multicellular horn, and Stephens 800X-2 network and attenuator. Range is from 30 to 18,000 cps. 25 watts power capacity. **Net $165.00.**

For an excellent three-way system, add a 214 super tweeter and 5000X network.



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