June 1956

Volume 1 Number 8

audiocraft

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

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

Robert A. Moog is an electrical engineer by profession and a musician by avocation. He became interested in the theremin six years ago; in 1954 he wrote an article on a simple version of the instrument. This elicited a great many letters from musicians who wanted him to make theremins for them, so he decided to design and manufacture the refined instrument described in this issue. It is the only standard product of the R. A. Moog Company (Flushing, N. Y.); other electronic instruments for musicians and audiophiles are made to customers' specifications.

About K. A. Alexander we know very little except that he resides in Chicago and that he has an engaging wit. His parody on Longfellow's "Blacksmith" is perceptive and filled with gentle humor, as you will discover on reaching page 22.

Omission from this issue of an article by Norman Crowhurst doesn't signify that the series on amplifier design has been discontinued. We simply didn't have space for it. The series will be resumed in the July issue, and will continue for some time.

Also upcoming is a construction article on a power supply for the amazing little TV amplifier described in AUDIOCRAFT for December, 1955, which will make it more universally useful. We've had many requests for this, and are happy to comply with them.

CHARLES FOWLER, *Publisher*

ROY F. ALLISON, *Editor*

FRANK R. WRIGHT, *Managing Editor*

ROY LINDSTROM, *Art Director*

ELEANOR GILCHRIST, *Art Assistant*

LUCIEN AIGNER, *Photography*

FRANCES A. NEWBURY *Manager, Books Division*

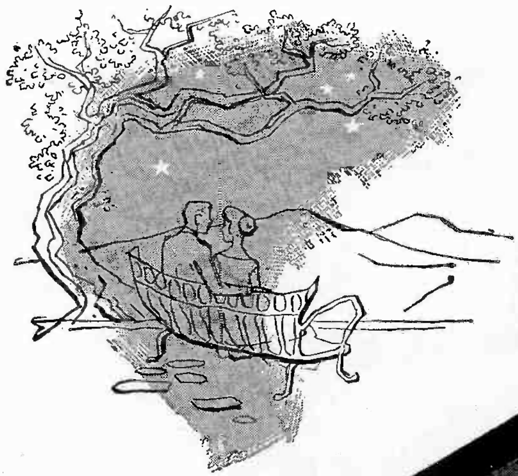
Contributing Editors
R. D. DARRELL
J. GORDON HOLT
JOSEPH MARSHALL

WARREN B. SYER, *Business Manager*

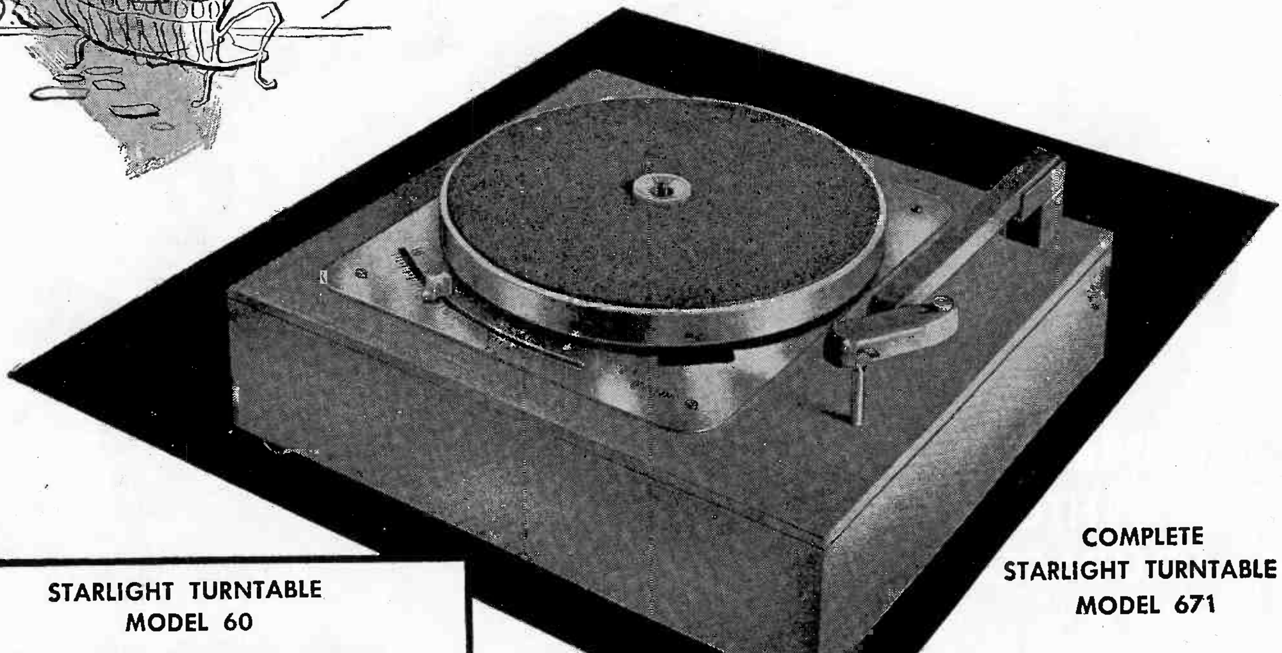
ARTHUR J. GRIFFIN, *Circulation Manager*

Branch Offices (Advertising only) — **New York:** John R. Rutherford and Associates, Inc., 30 Rockefeller Plaza. Telephone: Circle 5-4160. - **Chicago:** John R. Rutherford and Associates, 230 East Ohio St. Telephone: Whitehall 4-6715. - **Los Angeles:** 1052 West 6th Street. Telephone: Madison 6-1371. Edward Brand, West Coast Manager.

Audiocraft Magazine is published monthly by Audiocom, Inc., at Great Barrington, Mass. Telephone: Great Barrington 1300. Editorial, publication, and circulation offices at: The Publishing House, Great Barrington, Mass. Subscriptions: $3.50 per year in the United States and Canada. Single copies: 35 cents each. Editorial contributions will be welcomed by the editor. Payment for articles accepted will be arranged prior to publication. Unsolicited manuscripts should be accompanied by return postage. Entered as second-class matter October 1, 1955, at the post office, Great Barrington, Mass., under the act of March 3, 1879. Additional entry at the post office, Pittsfield, Mass. Printed in the U. S. A. by the Ben Franklin Press, Pittsfield, Mass.

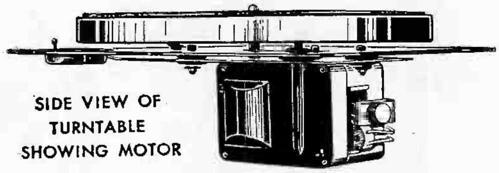


"as silent as the stars"



COMPLETE STARLIGHT TURNTABLE MODEL 671

STARLIGHT TURNTABLE MODEL 60



SIDE VIEW OF TURNTABLE SHOWING MOTOR

CHECK THESE IMPORTANT FEATURES

★ Continuous variable speed control from 16 to 83 RPM

★ Built-in illuminated stroboscope for setting speed exactly to 16⅔—33⅓—45—78.26 RPM

★ "wow and flutter" less than 0.2% RMS

★ Rumble and noise more than 40 db below average recording level

★ Massive, precision-machined aluminum turntable

★ Attractive, satin-finished ⅛" aluminum mounting plate

★ Over-all hum shield assures best performance with all types magnetic cartridges

★ Center drive • no belts • no pulleys • no cones

The METZNER Starlight HIGH FIDELITY TURNTABLE

Continuously Variable SPEED CONTROL and BUILT-IN STROBOSCOPE permits EXACT SETTING of 16, 33, 45, 78 RPM.

STARLIGHT TURNTABLE

MODEL 60 $49.50

(Mounting plate dimensions 12" x 13½".)

COMPLETE UNIT

Turntable, Pickup Arm, and Unfinished Birch Base

MODEL 671 $79.50

(Dimensions 16" x 17½" x 6½" overall.)

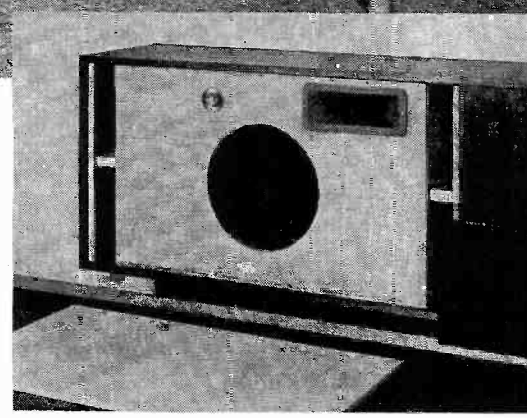
DEALERSHIPS AVAILABLE

Metzner ENGINEERING CORPORATION

1041 N. SYCAMORE AVE., HOLLYWOOD 38, CALIF.

Jensen *Authentic* HIGH FIDELITY LOUDSPEAKER KITS

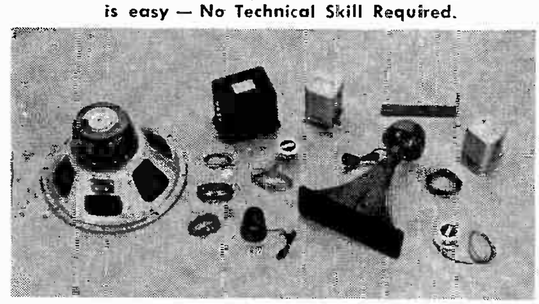
High Fidelity Installation For the Home by John and Jean Wehrheim, Architects.



Installation of Jensen Speaker System Kits is easy — No Technical Skill Required.

If you are planning to build-in your high fidelity system, you'll find a JENSEN Authentic High Fidelity Loudspeaker Kit not only gives superb performance but adds convenience and a note of simplicity to the whole project. Your builder or cabinet maker can easily follow the basic selected enclosure plan for the system you choose, (or do-it-yourself). New Manual 1060 tells all you need to know in simple terms, describes the eight JENSEN Loudspeaker Kits, and helps you make an intelligent choice of degree-of-performance.

Of course you can build your own free-standing speaker system if that is the arrangement you prefer. Again, Manual 1060 tells all. Why not order a copy today for only 50 cents.



Components For the Jensen KT-32 Triplex 3-way System Loudspeaker Kit.

Jensen MANUFACTURING COMPANY

Dept. N, 6601 S. Laramie, Chicago 38, Illinois

Division of the Muter Co. · In Canada, Copper Wire Products, Ltd. Licensee

WORLD'S QUALITY STANDARD FOR MORE THAN A QUARTER CENTURY

SEND FOR BIG 36 PAGE MANUAL FOR BUILDING YOUR OWN ENCLOSURES

Select the Kit and Cabinet you want from this complete manual of do-it-yourself designs. Simplified drawings, parts lists, speaker data—all planned to help you. Send 50¢ in coin for Manual 1060 today!

The Grounded Ear

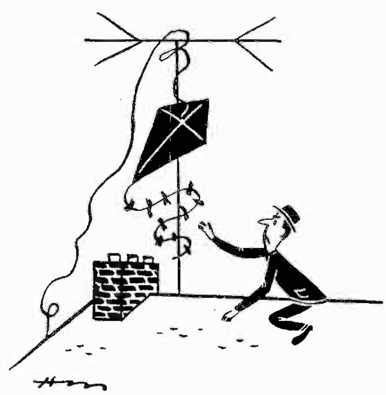
by Joseph Marshall

THE first commercial high fidelity equipment using transistors has now come on the market under the Fisher trade-mark. This is the new TR-1 preamplifier, equalized for the RIAA playback curve and with an input (so the advance publicity release said) suitable for all popular magnetic cartridges, even the low-level types which ordinarily require a transformer. I have not yet seen or used one, but the picture shows a very compact package about 4½ in. square and an inch or so high. It combines the transistor with the so-called printed-circuit type of assembly, and is powered by a self-contained battery.

The inevitable has thus occurred, and about on schedule. Unquestionably this will be followed by other high fidelity equipments using transistors instead of vacuum tubes. The transistor noise figure may not yet be comparable with that of the better tubes, but it is good enough to make the difference merely academic.

In practical circuits the over-all noise figure of the equipment can easily be superior when transistors are used, even though the noise figure of the transistor itself may not be as good. This assumes that, as in the Fisher preamp, a battery is used for the transistor power. In a previous note about the cascode preamp input circuit I pointed out that the noise level of a preamplifier is only partially a function of tube noise; it is determined primarily by hum from AC power supplies or an external AC field. Transistors have no filaments and this eliminates the most serious source of hum noise. A transistor may have twice as bad a noise figure as a tube, but if the tube filament is heated by AC, the resulting hum may be several times higher than the noise of the transistor. Furthermore, the higher noise figure of the transistor is not quite as disadvantageous in audio use as it would be in RF, because the output signal from most pickups is in the millivolt rather than the microvolt range, and is able to swamp the transistor noise more completely. Finally, the transistor is a low-impedance device; hum pick-up by induction is therefore less of a problem.

The low impedance of transistors is an advantage also, in that it permits dispensing with the input step-up transformers that are needed with low-impedance mikes or cartridges in vacuum-tube circuits. This, too, removes a source of hum that is difficult to minimize even when triple-shielded transformers are used. With battery power, the transistor may result in a considerable improvement in signal-to-noise ratio despite an inferior noise figure to begin with. To be sure, a good vacuum tube operated with batteries might be better than a transistor with batteries, but battery drain of tubes is too high to make the very slight improvement in noise figure worth while. The current drain of a transistor is so low, however, that battery replacement at long intervals should not be a great inconvenience.



It is now quite easy to obtain the 20 to 20,000-cps bandwidth needed to satisfy minimum hi-fi standards with current transistors, though rather heroic measures would be needed to obtain the extreme 10 to 100,000-cps bandwidth some people insist on even in preamps. Most of the audio transistor devices I have tried so far have had more distortion than good tube units, but I see no reason why distortion cannot be held down to whatever standards we want to set. The transistor in hi-fi is here to stay.

A pertinent question is, of course, what other kind of transistor devices we can expect. The phono preamp is a natural, because we are dealing with relatively low-impedance devices and because compactness fits in well here. The ideal place for a preamplifier-equalizer, so far as I'm concerned, is on the turntable board next to the pick-up arm, so that whatever adjustment needs to be made for a record can be made when it is put on the turntable. Transistor preamps can be made compact enough. Dissipating practically no heat, they can be hidden away in badly ventilated turntable bases, cabinets, odd corners of chassis, and so on. I think you can look for many more transistor-powered preamps.

Control units are the next most likely field. With transistors they can be more compact and less critical as to placement. The noise figure is less of a problem, since the input to the control unit approaches or exceeds 1 volt, even in the phono channel. There will be some delays because the low impedance of transistors requires the development of new tone-control circuits or modification of present ones. Also, until all preamps, tuners, and other devices that feed control units have low-impedance outputs, there will be some difficulty in the channel or source-switching stage. The biggest disadvantage at the moment is that transistors are single-amplifier devices, whereas tubes commonly used in control units are twin-triode types: it will take twice as many transistors as tubes to do the job, and this will run costs up. But all these are relatively minor problems, and the all-transistor control unit is inevitable.

How about power amplifiers? Unless miracles in transistors come upon us, this is a long way off. Present transistors deliver relatively low powers; on the other hand, the present trend is toward very high power outputs — the 50-watt amplifier is commonplace today. The transistor has one very basic advantage, however, which will accelerate the arrival of good high-power transistors: their low impedance might make it possible to dispense with the output transformer and couple directly to loudspeaker voice coils, thus realizing a dream which has driven some people to such extraordinary expedients as amplifiers which require a thousand watts input to deliver 10 watts output.

All of which means that those of you who like to tinker and experiment and have, like myself, bypassed the transistor because it was not quite up to hi-fi standards, ought to start looking into these gadgets pretty seriously and help expedite the inevitable.

Equalization Curves

The fact that the all-transistor Fisher preamp has RIAA compensation only gives me an opportunity to predict that this, too, is the start of an inevitable trend. I am aware that nowadays no control unit that does not offer at least 25 possible combinations of equalization will catch the eye of the cognoscenti, and there are a few which actually provide hundreds of curves. Offhand, there appears to be some justification for this variety for people who have large collections of records of a dozen or more labels acquired over the past 10 years. But even this is mainly a superficial impression. The fact is that there is no more difference between the *theoretical* recording curves of the RIAA and LP-NARTB-NAB family, or between the RIAA and the AES curves, than there is between two RIAA records on different labels, or even on the same labels. Actually, the RIAA playback curve will equalize either the old LP or AES curve within about 2 db throughout the 50 to 12,000-cps range. On the other hand it isn't likely, even if the same equipment is used all along the line in the recording process, that two RIAA recordings will match within less than 2 db. There are simply too many variables, beginning with the acoustic conditions of the hall or studio, to permit a higher uniformity. You might say that this in itself justifies the need for a wide variety of subtly different equalization curves, and that is certainly one point of view. But, I believe, this difference is so slight as to be difficult of discernment except to the trained or critical ear, and, in most cases, fully capable of compensation with the tone-control circuits in any event. No doubt the real crank (and I include myself in that class) will never want things any simpler than is avoidable, and will demand the most complete ability to compensate for every possible variation. But even we, when we settle down to *listening* to music instead of fiddling with controls, tend to establish standardized positions for controls and let them stay there. The less fussy music listener will be quite content with a single equalizer which will reproduce most records more faithfully, in unskilled hands, than an improperly used equalizer with too many options. In any event, I expect to see greater and greater use of the RIAA curve in new preamplifiers.

The Miratwin Cartridge

Some months ago I gave advance news of the coming of a new cartridge by the German manufacturer of Miraphon changers. This has now reached the

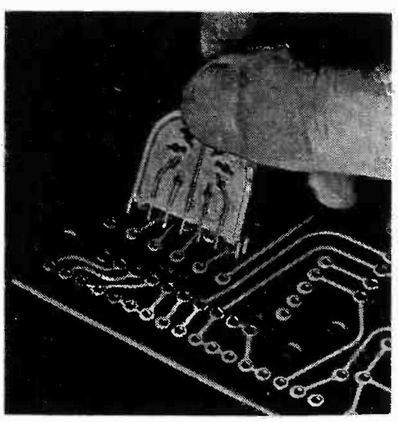
Continued on page 36

Audionews

RIGHT-ANGLE TUBE SOCKETS FOR PRINTED WIRING

Aerovox *Right-Angle Tube Sockets* are said to provide a marked reduction in the height and depth of printed-wiring assemblies. The sockets are claimed to be equally adaptable to hand- or machine-insertion methods. Terminals are of adequate length to slip into printed-wiring holes and be dip soldered. Silverplate contacts are designed to provide non-fatiguing contact pressure with suitable insertion and withdrawal pressures. Metal parts and mounting hardware are plated to meet salt-spray test specifications.

These components are available in 7- and 9-pin sockets and in four different versions: Type A for general-purpose applications where extra rigidity and resistance to vibration and shock are not important factors; Type AX for special applications requiring extra rigidity; Type B, same as A, but with a tube shield added; and Type BX, same as AX, but with a tube shield added.

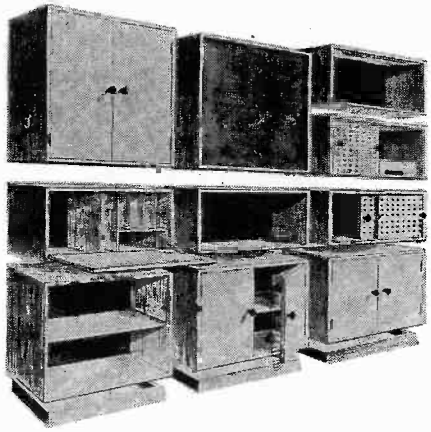


Aerovox tube socket for printed wiring.

CABINETS FOR HI-FI AND TV

A build-it-yourself home entertainment center for hi-fi and television, the *ST Series,* is Cabinart's newest addition to the cabinet kit line. Cabinart kits come ready to assemble (with only a screwdriver) or factory assembled, ready to finish or paint.

Ten units, nine cabinets and one base, comprise the ST Series. Cabinets are modular and may be purchased singly allowing the hi-fi enthusiast to compose and select the wall-storage combination most suitable. Cabinets are identical in width. Heights of 16, 24, and 32 in. are available, while a 16-inch depth permits all-purpose storage in every cabinet.



Cabinart's home entertainment center.

A basic starter combination is the Equipment Cabinet, Model ST-9 or KST-9 ("K" denotes "kit") and the Bass Reflex Speaker Cabinet, ST-5 or KST-5. High fidelity radio tuners and amplifiers can be mounted in one section of the equipment cabinet. A slide-mounted phonograph drawer will accommodate any hi-fi record changer or player in the other compartment. Other cabinets may be added later for tape recorders, television sets, phonograph record storage, and so on.

Cabinet doors and moldings are of birch. Cabinet shells are constructed of heavy, ¾-inch, white-pine plywood. Prices begin at $21.00, retail. Hardware, and assembly and finishing instructions, are furnished with each unit in the series.

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

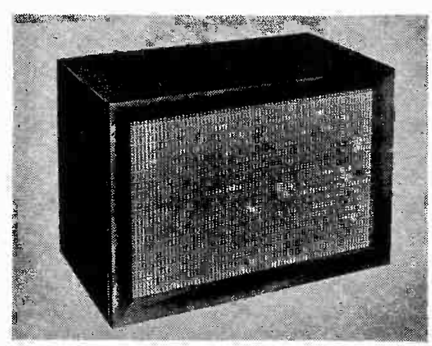
"KARLSON 8" SPEAKER ENCLOSURE

A new line of loudspeaker enclosures utilizing the Karlson principle has been introduced by Karlson Associates, Inc. These enclosures are designed specifically for use with eight-inch speakers.

There are five different models of this new enclosure available with the same size and construction, differing only in the degree of finish. The basic unit of the new line is an easy-to-assemble kit. Next there is available a factory assembled kit, then a painted unit finished with three-color paints. The de luxe models are finished in blond or mahogany plastic.

The Karlson 8 kit comes complete with all necessary parts precut and can be assembled easily. Only a hammer and glue are necessary for the assembly since all the parts, hardware, and grille cloth are included in the kit.

The Karlson 8 is designed for installations where space is limited or where portability is of importance. Complete with speaker, the unit weighs only about 15 pounds and is 17¼ in. by 11¾ in. by 10 in.



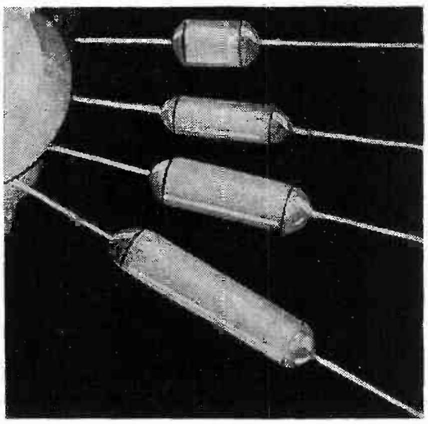
Karlson 8 is available in five models.

Included with each Karlson 8 is a booklet explaining the basic theory of the acoustics of the system. The booklet also goes into detail on the larger Karlson 12 and 15 enclosures, explaining the construction and uses of these units.

A copy of this 32-page booklet may be obtained free from Karlson Associates, Inc. Ask for booklet 132.

SUBMINIATURE ELECTROLYTIC CAPACITOR

Designed specifically to meet the unique electrical requirements and dimensional limitations of extremely small equipment, a new subminiature series of



New midget tantalum electrolytics.

Tantalum capacitors, *Type NT,* has just been developed by the Cornell-Dubilier Electric Corporation.

These new subminiature capacitors are available in a wide selection of ratings and sizes down to the most minute dimensions. Ratings range from 0.5 volts DC to 16 volts DC working. Capacitances range from .08 to 30 μfd, depending on voltage. Case sizes are only 3/32 in. or 1/8 in. in diameter, and 5/32 in. to 1/2 in. in length.

Type NT capacitors are suited for tight-fitting applications where high stability and other favorable electrical characteristics plus minimum size are essential. These applications include transistor circuits for hearing aids and miniature radio receivers, printed circuit assemblies, subminiature controls, and other very small, low-voltage devices designed for operation within a temperature range of from —20° to +55° C.

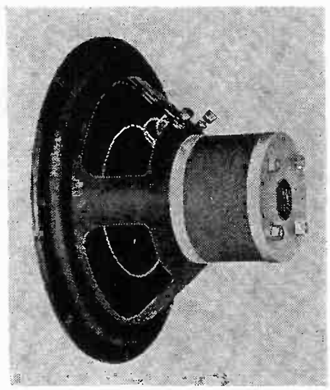
Superior capacitance, stability, and power-factor characteristics are said to be assured by a special-formulation neutral electrolyte. The tantalum wire anode is encased in a porous, absorbent sleeve which serves as a separator. It terminates in a nickel anode lead, tinned for easy soldering in making capacitor connections. The solid silver tubular outer case serves as the capacitor cathode.

Type NT subminiature capacitors are hermetically sealed. Additional features are low leakage and long shelf life. They are said to provide exceptionally long service and highly dependable performance under all conditions within the specified temperature range and electrical ratings.

NEW HARTLEY LOUDSPEAKER

Recently announced is the improved *Hartley 215* loudspeaker. It is claimed

Sectioned-cone Hartley 215.

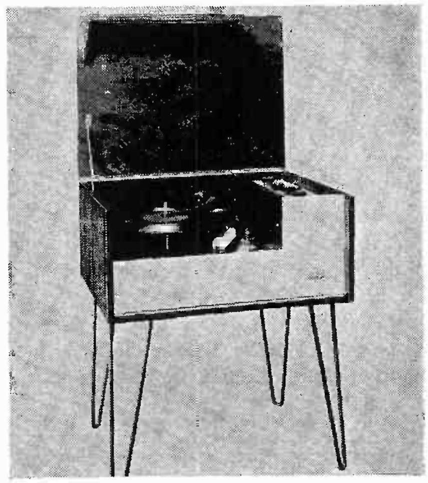


to have no resonance through a response range of 1 to 18,000 cps. With a five pound magnet and peak power of 20 watts, the loudspeaker employs one cone, sectioned, with a mechanical crossover that is said to avoid any break in tone. Cloth suspension is reported to allow cone movement of 1/2 in.

The improved 215 is priced at $65.00.

ALTEC LANSING RECORD REPRODUCER

The 901A *Melodist* Record Reproducer



Hi-fi package unit by Altec-Lansing.

is a new entry in the list of Altec Lansing's high fidelity products.

The unit, available in either blond or mahogany cabinet, contains Altec's A-339A ten-watt amplifier and control preamp, a Collaro three-speed changer, and a dual stylus GE variable reluctance cartridge. The 901A is available with or without legs. Consumer price is $225.00.

Appropriate Settings: No. 3

MODEL April Stride is shown here listening to the new James B. Lansing *Jourdan* corner reflex enclosure.

The Jourdan, first shown at the Hi-Fi Music Show in Los Angeles last February, was selected by the Herman Miller Furniture Company as a high fidelity enclosure with a "fine furniture" look. In this setting it looks fine to us too.

The enclosure can be fitted with an 8-, 12-, or 15-inch speaker, or the .001 or .002 (network) system. Further information about the Jourdan enclosure will be furnished on request.

TIPS FOR THE WOODCRAFTER

by George Bowe

Basic Power Tools, Part I

In earlier issues we discussed basic hand tools and how to use them. Starting this month, let's explore the realm of power tools where, with less effort, faster production and better craftsmanship can be yours. Perhaps you have hesitated to purchase motorized equipment because of physical risk. With the ownership of any power tool goes the responsibility of safeguarding yourself and others in the vicinity of the tool, but the safety rules are common-sense rules that anyone can learn and practice. Before we get into a discussion of a specific piece of equipment, let's examine some safety practices that apply to all power tools:

1) Stationary power tools should be securely mounted on suitable benches or stands. When using portable tools make sure the work is held securely in place.

2) Make sure you know how to use the machine. Do not try to use it until you understand it thoroughly.

3) Wear snugly fitting, short-sleeved clothing with no loose parts. Neckties are dangerous — even rings and wrist watches are hazards.

4) Keep the area around the machine clean. Give yourself plenty of light and room to work in. Remove any obstruction to free movement.

5) Always keep your eyes on the work and your fingers away from rotating parts.

6) Always use sharp tools for better and safer work with least effort. The added force necessary to make a dull tool cut can cause a dangerous slip or glance-off.

7) Sawdust and chips should never be brushed or blown off a power tool while it's running.

8) Always turn off the power and wait until all moving parts stop before adjusting or oiling the machine.

9) Switches should be located conveniently to the operator so that reaching isn't necessary.

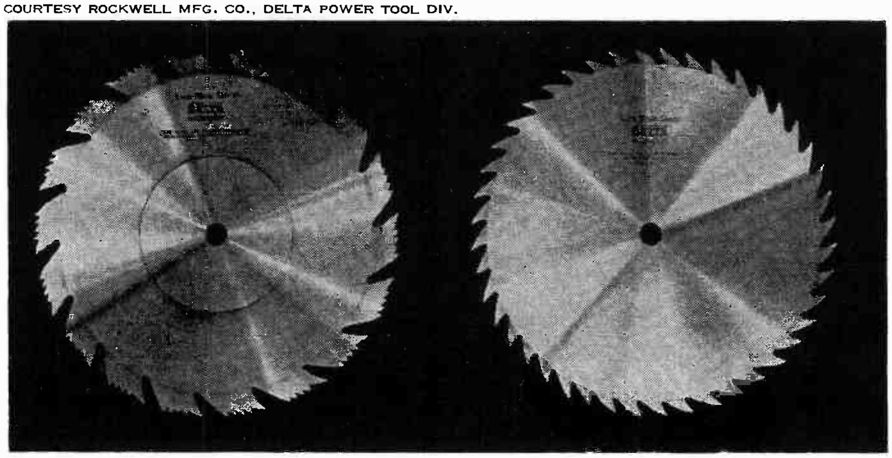
10) Keep children away from power tools. If possible, in the workshop have a master switch that can be locked.

It doesn't take very long for these safe practices to become second nature to a power-tool operator, enabling him to master the operation of the machine and get the most out of it.

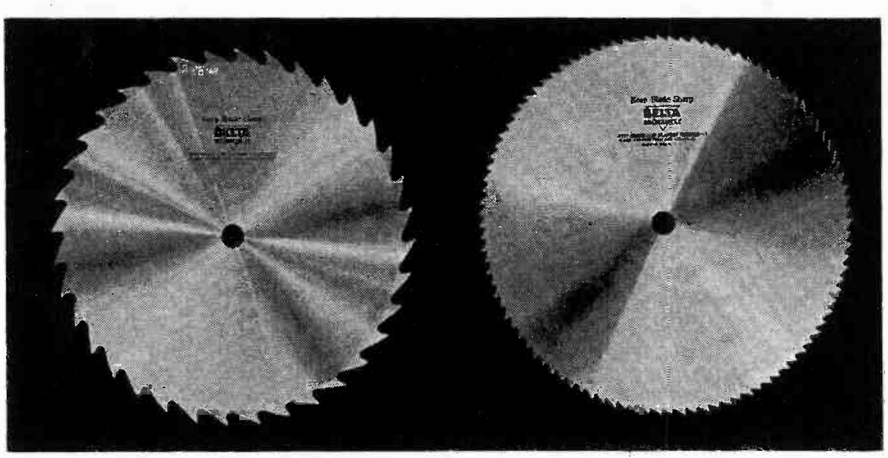
The Circular Saw

For our first discussion we've selected the power tool that is the initial choice of most home craftsmen — the circular saw, sometimes called table saw or bench saw. The great versatility of this machine makes it a "must" in the workshop for many simple and intricate cutting operations. The name is derived from the shape of the blade, and the diameter of the blade indicates the size of the saw. For amateur use the blade sizes range from 6 in. to 10 in., the most popular for home use being the 8-inch saw. Although circular saws are made in a variety of styles, the tilt-table and tilt-arbor are the most common. With the former, it is necessary to tilt the saw table and the stock you're working on in order to make an angle cut. With the tilt-arbor saw, the blade pivots and locks at the desired angle while the stock remains level on the table. The latter is more practical and less dangerous.

COURTESY ROCKWELL MFG. CO., DELTA POWER TOOL DIV.



Above: hollow-ground blade for circular saw. Far right: combination blade, most popular in home shop. Below: crosscut blade. Far left: ripping blade.



COURTESY ROCKWELL MFG. CO., DELTA POWER TOOL DIV.

There are literally hundreds of types of circular-saw blades designed for specialized use, but for the average home craftsman four basic blades will suffice.

1) *Crosscut* or *cutoff saw*: the teeth are similar to those of a hand crosscut saw. Its primary work is to square and trim stock.

2) *Hollow-ground saw*: this blade makes a very smooth cut similar to that of a plane. It is used for fine cabinetwork where edges must be cut with extreme accuracy.

3) *Rip saw*: this has chisel teeth and is used for ripping (cutting with the grain of the wood).

4) *Combination saw*: this blade has a combination of crosscut and rip-saw

teeth enabling it to cut both across the grain and with it.

While these blades handle the basic cutting operations, there are a number of accessories available for performing specialized functions. One that finds considerable application in general cabinetwork is the dado head, which is actually a group of blades. Its principal use is to cut channels in wood. It consists of two identical combination blades with several cutters or chipper blades to be sandwiched between them, depending upon the desired width of the cut. It can cut a rabbet or a dado as wide as 13/16 in. (A rabbet is a groove made with the grain; a dado is a groove made across the grain.) It can cut blind and corner dadoes, and can

COURTESY ROCKWELL MFG. CO., DELTA POWER TOOL DIV.

Component parts of dado head assembly.

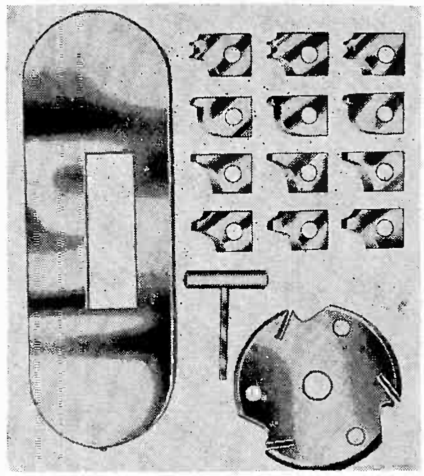
turn out various joints such as end lap, half lap, cross lap, edge lap, and middle lap.

Moldings of almost unlimited design can be produced by the circular saw if it is equipped with a molding head. There are many different types made by various manufacturers, but essentially it is a cutterhead which replaces the regular blade. Into the rim of the cutterhead a variety of cutter knives can be fastened with screws. The choice of knives is determined by the shape of molding desired. While a molding head does a fair enough job, it cannot be presumed to take the place of a shaper, which is a machine especially designed for such work. In turning out moldings, the secret to a good cutting job is the high speed of the machine. The shaper is built for speed and specifically for this type of work. The circular saw with a molding head does not have the speed nor the safety features of a shaper; but when a shaper is not available, the saw is a welcome pinch hitter.

There are individual blades of extra thickness — 1/4, 5/16, or 3/8 in. — known as groovers. As the name implies, they make grooves of those particular widths and are excellent for making a tenon for a mortise-and-tenon joint. For instance, if the stock is ¾ in. thick, running each side through the ¼-inch groover in one operation would leave a ¼-inch tenon in the center.

By using an abrasive cutting wheel, the circular saw will cut metals and sharpen tools. However, in this type of operation there is always the chance

COURTESY ROCKWELL MFG. CO. DELTA POWER TOOL DIV.



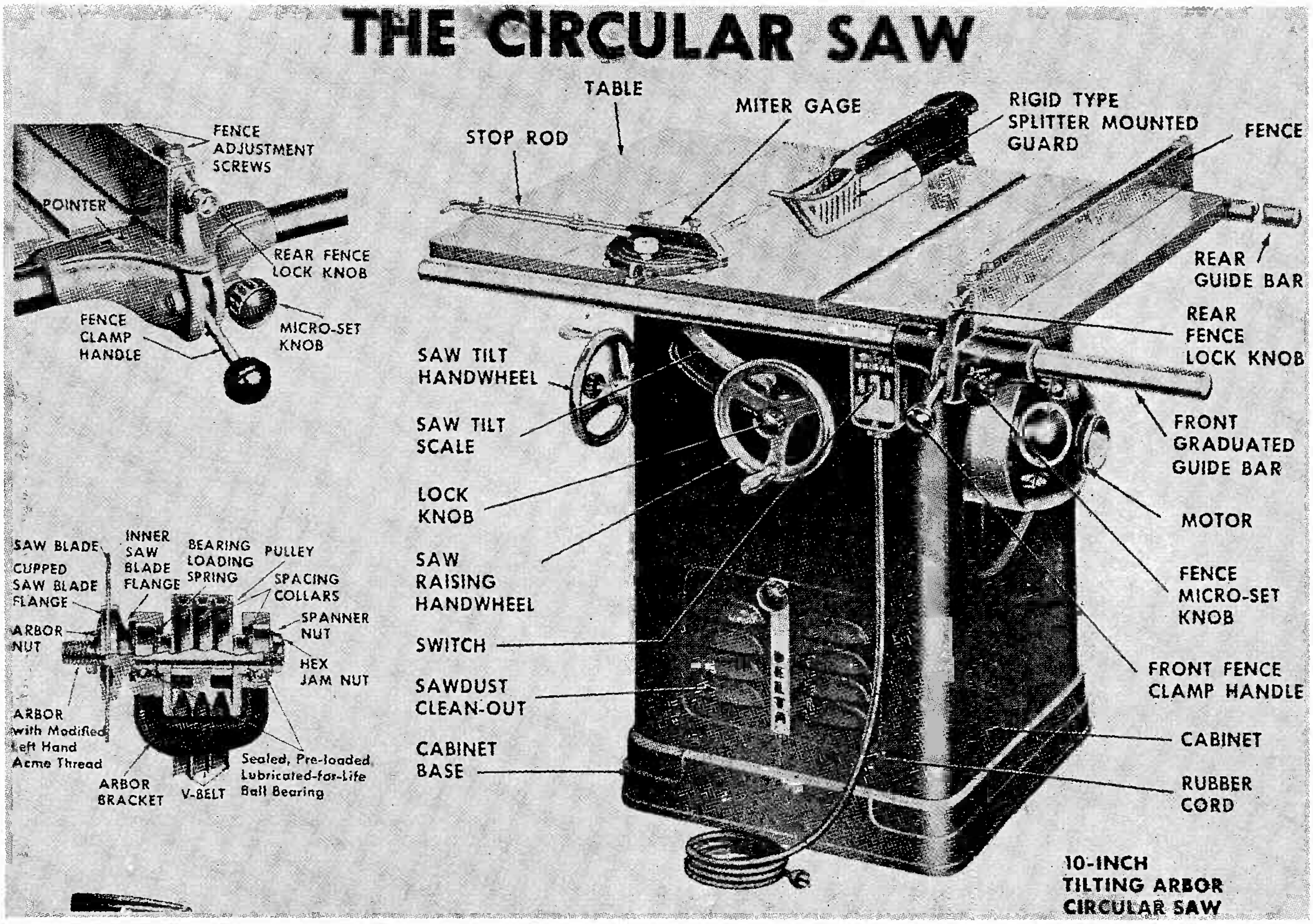
Molding cutterhead knives, table insert.

that grit from the abrasive will find its way into the bearings of the machine and cause damage. Perhaps the best rule is to confine the saw to the material for which it was originally designed — wood.

Of the various components of the tilting arbor saw there are four which receive the most constant use. The *blade-raising handle* usually is a wheel or lever on the front of the machine to raise or lower the saw blade, the height being indicated by a scale with a movable pointer. One of the most

Continued on page 35

A 10-inch power saw. Some units have all the parts shown here; in others, accessories such as base and saw guard are extra.



COURTESY ROCKWELL MFG. CO., DELTA POWER TOOL DIV.

SOUND SERVICING

by Irving M. Fried

Loudspeakers, Part I

Almost every owner of high fidelity equipment is unhappy with his loudspeaker system. Surveys indicate that at any one time over 60% of hi-fi'ers plan to make radical changes in their speaker systems. Common complaints about speakers range from too much bass (or too little), too much treble (or too little), through the range of other particular faults on to the vague objection, "I don't enjoy listening to it."

Whatever may be the cause, it is apparent that only a complete change in equipment will make most audiophiles happy. It is also obvious, however, that their "next" speaker will be 1) carefully planned, 2) easier to listen to, and

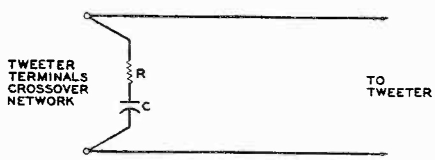


Fig. 1. Peak-suppressor for a tweeter.

3) quite expensive. While the planning is going on, the typical listener might like to experiment with his present equipment in an attempt to alleviate the irritants and distortions there present. The day of reckoning might even be put off.

Nearly everything that can be wrong in a loudspeaker is summed up by the word "distortion". Distortion of some sort is present in every loudspeaker — harmonic, doppler, intermodulation, phase, frequency, transient (including "tone burst"), and others. One factor is common to all these distortions: they are all representative of some kind of non-linearity, and nearly every kind of distortion is accompanied by peaks and valleys in frequency response. Conversely, the smoother a speaker system becomes over a certain pass band and at a certain power rating, the less distorted the total sound will be.

Therefore, anything the amateur audiophile can do to smooth out his speaker response will return dividends in terms of less distortion and less listening fatigue. Let us work on this idea for a while, and first of all on that ornery mechanism known as the tweeter.

Most tweeters have too much crackle and hiss, no matter what the program material. These are not so much symptoms of wide range as of high distortion. Then, any means of cutting down on the crackling and hissing (any way of reducing response irregularities), even if it means restricting the frequency range slightly, will make for lower distortion and better listening.

One of the older remedies for peaky tweeters is to connect a large resistor across the tweeter (10 to 20 watts is recommended), whose value is between one and two times the nominal impedance of the tweeter. The resistor cuts down the tweeter level, and seems to "bleed" the peaks — when the tweeter impedance rises (on peaks), the fixed resistance absorbs a greater proportion of the power, which reduces the amplitude of the peaks. It also furnishes a more nearly constant load for the cross-over network which makes it work under more nearly ideal conditions.

Another technique, recommended and used by several manufacturers, is to wire across the high-frequency output terminals of the crossover network (which is the same as across the tweeter) a small resistor in series with a capacitor, Fig. 1. The latter should be of such value that its impedance becomes insignificant compared to that of the resistor at a suitable high frequency. The method seems to help by smoothing out peaks, just as the resistor alone does, and rolling down the highs progressively with frequency — a technique that will always tend to make the over-all sound seem smoother.

Just what values of R and C you should use is a matter for cut-and-try determination. Try first an R of a very few ohms, say 1/5 the tweeter impedance, and a C which will be a very low impedance at 20 Kc.

Other methods of limiting tweeter power are the use of L or T pads, and sometimes even disconnecting the tweeter. Many people simply like a tweeter that operates at a lower level.

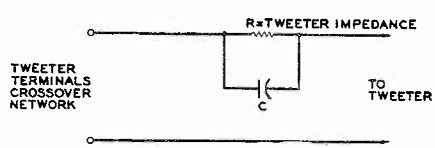


Fig. 2. High-frequency response booster.

If you are one of them, you may want to reduce the level. Then, too, if you play many older, scratched LP's or 78's, you may find that you get better results by disconnecting the tweeter. In effect, this is putting in a low-pass filter, in the accepted traditional method.

There are still other possibilities for improving performance. Among these are raising the crossover point. In certain de luxe speaker systems inexpensive tweeters are used with a crossover network that operates an octave above the specified tweeter cutoff point, with uniformly better results. The probable reason for this is that most manufacturers of tweeters tend to be a bit too enthusiastic in their claims of low-frequency cutoff. If you move the crossover up, you let the tweeter work in a more suitable range for it. Try reducing the capacitor in the crossover network, the one that is in series with the tweeter tap, to ½ its value. That raises the crossover an octave. Be sure, when you do this, that the woofer or mid-range unit just below is allowed to roll off naturally (rather than being cut off by the network), else you may put a big hole in the response.

Then, too, (but only after the crossover point has been pushed up), you may want to tip up the extreme highs of your tweeter. If you have efficiency to waste, you can install a losser network that will reduce the low-end efficiency of the tweeter in favor of the very top. Try inserting, *in series* with your tweeter, a resistor of the same value of resistance as the nominal impedance. In parallel with this resistance, Fig. 2, solder a capacitor of such value that its impedance is equal to that of the resistor at the frequency for which you want the boost to begin (say 5 or 7 Kc).

You may find that, for the first time, your tweeter reproduces the breathy sounds that denote ultra-wide range, rather than the shrieky sounds that denote too many middle-range peaks.

Also, you might try redirecting your tweeter. Many home experimenters find that the over-all effect is improved merely by directing the tweeter into a corner, or up to the ceiling, or up and at a diffusing board suspended 45° from horizontal. With such improved diffusion, many tweeters that sound narrow and scratchy head-on begin to sound sweet and good. Indeed, several of the English speaker manufacturers seem to feel that any tweeter is better if listened to after its narrow beam has been broken up by some diffusion mechanism.

If you can't alter your tweeter position, you might want to try other types of diffusers. One possibility is a piece of cardboard, bent in the shape of a V, fastened in front of or behind the grille cloth. Another method is to use a piece of hard material, shaped like half an egg shell, inverted and fastened to the grille cloth in front of the tweeter.

One final method: judicious speaker placement. I have found that speakers at ear level always sound brighter than when placed above or below. I have found also that peaky-sounding speakers

Continued on page 36

TAPE NEWS & VIEWS

by J. Gordon Holt

Recording from Radio

At least two booklets that have appeared on the market in recent years have listed 1,001 uses for the tape recorder.

The listed applications range from recording orchestra performances for posterity to recording body sounds for scientific study. About midway down the list is off-the-air recording of news events or musical concerts, for any purpose the recordist sees fit. (The fact that these recordings *can* be used as the solid basis for an illicit record company is generally overlooked, although it has struck me that the authors of these booklets should mention the matter, at least, in connection with a rough estimate of the usual penalties for taking such liberties.)

The tape recorder is unequalled as a preserver of radio broadcasts, because it is easy to connect and easy to use. The resulting tapes can be edited to remove extraneous coughs and commercials, while, if the whole thing proves to be unworthy of indefinite preservation, it can be erased and the tape used over again.

To the man who doesn't mind pulling plugs out and inserting them into different sockets each time he wants to record or listen to his tuner or recorder, the procedure for piping the signal from his tuner into the tape recorder is extremely simple. He just unplugs the tuner from the control unit and puts it into the high-level input of the tape recorder. Then, he connects a cable from his tape recorder's output back into his control unit, and he's all ready to record and to listen to what he's recording. Since all tape recorders worth using with a high fidelity system have some form of output connection, this can be used for feeding back into the control unit, and if the recorder has a high-level high-impedance output preceding its power-output stage, so much the better. (The installation of a high-impedance output connection from a recorder was described in some detail in "Tape News and Views" for January 1956.) A plugged-in connection for a recorder is shown in the block diagram, Fig. 1.

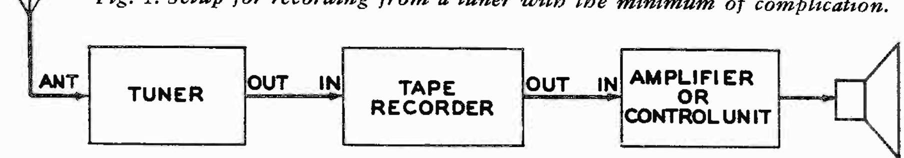


Fig. 1. Setup for recording from a tuner with the minimum of complication.

Not everyone, though, likes to be constantly yanking and inserting plugs. The tendency is, in fact, for the conscientious high fidelity enthusiast to try to get his system's operation as slick as possible, using switches to perform all the input and output connecting operations.

A system chosen with an eye to off-the-air recording will have an auxiliary tape output coming from either the tuner or the control unit, and will have at least enough high-level input connections to the control unit to allow the recorder's output to be left connected at all times. Connection of the tape recorder into such a system is shown in Fig. 2. The input to the recorder is fed by the TAPE OUT connection from the tuner or control unit, and the tape's output is fed into one of the high-level inputs on the control unit.

When recording off the air with this setup, the control unit's switch position will depend upon whether the TAPE OUT connection is coming from the control unit or directly from the tuner. Fig. 2 shows the control unit's TAPE OUT connection being used, so the input selector switch for this arrangement should be set to TUNER for recording. The rest of the system will then operate as it normally would if the tuner were being used by itself; the recorder is simply bridged across the program chain, pulling the signal out wherever convenient. The program may be heard at all times, and the recorder may be started whenever desired.

In playback, the selector switch is turned to TAPE, and the tape's output is piped through the system.

The alternative connection, with the tape recorder being fed by a TAPE OUT connection directly from the tuner, allows the input selector switch to be set in either the TUNER or TAPE position while recording. In the TUNER position, the hookup is the same as before except that the recorder is bridged across an earlier point in the circuit. In the TAPE position, the tuner's signal is still heard, but this time it is heard after it has passed through the recorder's amplifier. With the average tape recorder it doesn't matter much which position of the input selector is used for monitoring; the signal is heard before it actually gets onto the tape anyway. But with a recorder that uses a separate playback head, so that one can monitor directly *from* the tape while recording, the use of the TAPE position on the selector is a definite advantage. With this type of recorder, the TAPE OUT connection from the tuner may feed the recorder directly as before, but the selector switch set to TAPE actually inserts the entire recorder in series with the system. When the recorder's output selector knob is set for playback while *recording*, the signal heard is the playback of the tape—assurance that the recording is getting onto the tape.

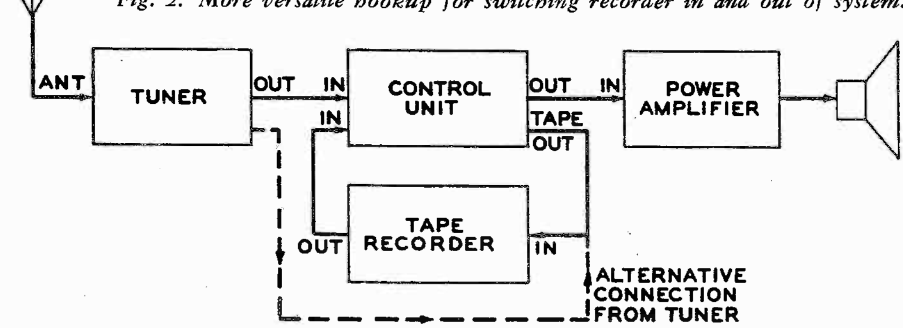


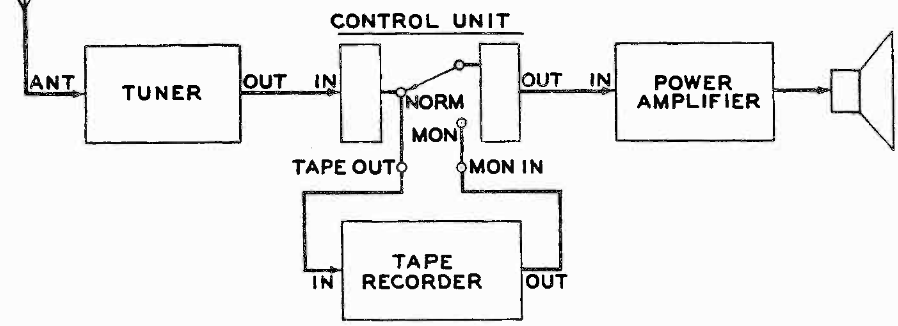
Fig. 2. More versatile hookup for switching recorder in and out of system.

Some control units have, besides the now-standard TAPE OUT connection, a separate tape input marked MONITOR, plus switching facilities to break the signal circuit at that point. The three-headed tape recorder may then be connected to these input and output receptacles so that, with the control unit's switch set to MONITOR, the recorder is inserted in series with the signal circuit (Fig. 3). With the switch set for NORMAL, the recorder is bypassed and the system is set up for usual operation.

procedure for a broadcasting station (particularly a "good-music" station) to set the volume of speech well below that of music, a preceding speech program should be leveled in at about half of normal recording volume. With a recorder having a neon-bulb indicator, or two neon bulbs, this means that the level for a speech program should be set below the point at which it causes the NORMAL bulb to flash.

If the program preceding the desired one consists of musical material, the

Fig. 3. Still more elaborate system permits monitoring from tape recorder.



At this point, a few recorder or control-unit deviations that can disrupt these nicely organized procedures might be discussed briefly.

There are some control units that have their input-selector-switch circuits worked out so that all inputs that aren't being used at the moment are shorted to ground, to prevent them from leaking through into the background of the active channel. If the tape recorder is being used in an arrangement such as that in Fig. 2, and has its recording-level indicator connected across its output (as in the Ampex units, for example), the signal will be blanked out by the grounded control-unit input, and no reading will show on the indicator. This combination of recorder and control unit must either be used with an external switch between the recorder output and control-unit input, or the recorder must be fed by an auxiliary output from the tuner — the "alternative connection" mentioned earlier. If the tuner does not have a second output, one can be added very simply just by drilling a second hole in the tuner chassis beside the regular output, and paralleling the connections of the output receptacles.

Off-the-air recording procedure is very similar to disc duplicating, except that the recording must be done at a time chosen by the broadcasting station rather than at the convenience of the recordist.

The preliminaries involve tuning in the desired station at least 15 minutes (preferably a half hour) before the recording project, to allow the tuner to warm up beyond its drift period, and setting the record volume to that of a preceding program. Since it is normal

matter is somewhat simplified. This can be set for the normal level that will be used during the recording session, with the recording indicator showing full level on the highest peaks.

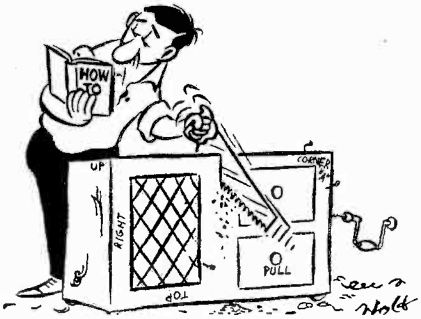
At the moment the program is about to begin, the recorder should be started in sufficient time to get the opening section of the program. It's better to get too much of the preliminaries on the tape than to miss part of the desired program, and any excess can be erased or edited out at a later time.

A word about playing time. Most musical programs of any great worth run from a half hour to an hour or more in duration. If the recorder's maximum capacity is a 7-inch reel it is advisable to use extra-play or double-play tape, even if half the reel never gets to be used. The listed program schedule is usually a pretty good indication of how much tape should be put onto the recorder, since the chances of a program's running overtime into another sponsor's allotment are practically nil in these days of counted and accounted seconds.

Once the recording is begun, the job consists largely of watching to make sure that the recording level is satisfactory. Remember, though, that if an unexpected volume peak comes along, the engineer at the station's control panel is likely to take things into his own hands and turn it down. If the recorder operator decides to whip the volume down at the same time, the total of the two attacks on the volume will be very conspicuous as a sudden collapse of the signal. So before pulling the volume down below the level determined by

Continued on page 33

Audiophile's Bookshelf



Order your audio and do-it-yourself books directly from us, using the convenient order blank. We make every attempt to take care of your order by return mail.

JUST FILL IN THE COUPON BELOW

EQUIPMENT

YOUR TAPE RECORDER, Robert and Mary Marshall. The first book, for nonprofessionals, devoted exclusively to the tape recorder. Gives the complete story of what it consists of and how to use it. Based on more than 2500 experiments. Amply illustrated.
No. 202 .... $4.95

HOW TO INSTALL TV ANTENNAS, Samuel L. Marshall. A completely practical, illustrated "Antenna Bible". Tells you everything you need to know about installing TV antennas: safety precautions, putting up masts and towers, getting the best reception in fringe areas, etc. In short — how to do the job RIGHT . . quickly, safely, economically.
No. 162 .... $2.50

HI-FI LOUDSPEAKERS AND ENCLOSURES, Abraham B. Cohen. A complete, well-written book dealing with one of the most important features of a hi-fi system. Includes an appendix of 18 complete plans for construction.
No. 209 .... $4.60

DO-IT-YOURSELF

BUILD IT YOURSELF — 25 furniture designs. Specifically prepared working drawings. Clear and easy-to-follow instructions for making colonial, modern and contemporary furniture. 64 pp. illustrated.
No. 158 .... $2.15

IMPROVE YOUR HOME WITH BUILT-INS, Robert Schraff. How to make scores of built-ins for rewarding, low-cost home improvements. Over 200 plans for easy-to-build, space-saving units for every room in the house. The book takes you every step of the way.
No. 159 .... $4.95

POWER TOOLS FOR THE HOME CRAFTSMAN, Edwin G. Hamilton. Helps you do more kinds of jobs . . . produce better results — faster — easier. Start right with new tools, save time, money. Home craftsmen, planning to build their own speaker cabinets, will benefit from this handy, practical manual.
No. 160 .... $4.95

BASIC TOOLS FOR WOODWORKING, (2nd edition) Graphic illustrations and brief explanatory text indicate the right and wrong ways of using all carpentry tools. Basic, reliable information and guidance. 136 pp. illus.
No. 161 .... $3.25

BINDERS FOR AUDIOCRAFT

Now available — for only $3.50 each. Attractive, protective blue leatherette, to hold 14 issues of AUDIOCRAFT. Use coupon below to order.

MISCELLANEOUS

The New **HIGH FIDELITY HANDBOOK,** Irving Greene and James Radcliffe.
Introduction by Deems Taylor. A complete, practical guide for purchase, assembly, installation, maintenance and enjoyment of high fidelity music systems. 250 illustrations, diagrams and plans.
No. 200 .... $4.95

HOME MUSIC SYSTEMS (Revised Edition) Edward Tatnall Canby. 302 pages, illus. This popular guide to high fidelity has been completely revised. Explains the operation of a radio-phonograph, where to buy separate parts, and how to house them. One chapter devoted to suggested combinations of equipment.
No. 151 .... $3.95

THE HIGH FIDELITY READER, edited by Roy H. Hoopes, Jr. Introduction by John M. Conly. An anthology of outstanding articles originally appearing in HIGH FIDELITY Magazine covering various aspects of the high fidelity phenomenon.
No. 155 .... $3.50

HIGH FIDELITY RECORD ANNUAL — A first volume of record reviews — classical music and the spoken word — from HIGH FIDELITY Magazine. Edited by Roland Gelatt.
No. 201 .... $4.95

BUILDING YOUR RECORD LIBRARY, Edited by Roy H. Hoopes, Jr. An excellent guide to more listening enjoyment. 17 qualified experts help you to build a well-balanced record library, custom-tailored to *your* individual taste.
No. 208 .... $3.95

Book Department
AUDIOCRAFT Magazine
Great Barrington, Mass.

I enclose $.................... for which please send me, postpaid, the books indicated by the circled numbers below. (No C.O.D.'s or charge orders, please.)
Foreign orders sent at buyer's risk. Add 55¢ for postage on foreign orders. Binder #1 (14 issues)

151 154 155 158 159 160 161 162 200 201 202 208 209

NAME ....
ADDRESS ....
CITY .... ZONE .... STATE ....

READERS' FORUM

Gentlemen:

I wonder if something couldn't be done in the way of an article on how to improve your present amplifier — Williamson or adaptation of same — without spending money for a new transformer.

All of the present articles on "how to" are pointed toward purchasing a new transformer.

How about the thousands having a transformer similar to an Acrosound TO-300 or actually an Acrosound TO-300? I'm sure there are many new ideas that would improve these amplifiers: for instance, the new Hafler front end, or perhaps the front end recently espoused by the Heath Company, or the slight changes to improve stability talked about in your January issue.

Many of these changes or improvements I'm sure can be used successfully with the Acrosound T-300 or equivalent.

It is unfair to assume that everyone who reads your magazine is either an electronics engineer or an electronics serviceman. I'm sure a poll would find the greatest number are just Joes who have enough basic knowledge to get by and who can read a diagram properly.

So I'll be looking forward to an article on these circuits and circuit changes that don't require going out and spending another \$30 or \$40. As it is, the articles and ads keep me broke and I'm sure many of my fellow readers as well.

A. J. Sterbenz
Cleveland, Ohio

As Mr. Hafler pointed out in his article, modification or modernization of the Williamson amplifier involves two distinct steps: first, increasing the margin of stability; and second, increasing the output power above that of the ultra-linear connection.

All the information necessary to accomplish the first objective is given in this article. It isn't necessary at all to increase the output power or make any changes in the output stage in order to accomplish the increased stability.

In order to use different output tubes of greater power capacity, a new output transformer must be added at the same time because the output transformer determines the maximum power as much as the tube type does. To get 50 clean watts from an amplifier you simply must have a transformer capable of handling this, no matter which output tubes you use. — ED.

EDITORIAL

WE have been sounding off rather tediously, it seems to us on rereading previous editorials, about the shabby treatment of FM at the hands of those who could use it kindly *and* profitably, and of the dangers it faces from outside interests. It is therefore pleasant to be able to report a most encouraging development. A major advertising agency (Maxon, Inc.) has released a comprehensive report on a thorough study of FM broadcasting it has just completed. This report, entitled "FM Radio: The Frustrated Medium", does little more than confirm what FM people have been saying for a long time — but it will carry vastly greater weight, because it comes from an obviously disinterested but highly respected authority. In short, it cannot be ignored, either by other agencies or by advertisers.

As the report itself points out, "How fast or how much [FM continues to grow] depends to a very great extent on the interest of the major national advertisers." Once this interest is aroused, and some intelligent experiments made, FM should be on the way toward its rightful place as the most important sound-broadcasting medium. And the Maxon report must arouse a substantial amount of interest.

Here are FM's advantages, as listed in the report, over AM and TV broadcasting: 1) lower costs to advertisers; 2) consistently better reception; 3) coverage of the same area day or night and in any season; 4) an audience with higher average income; 5) a greater proportion of adult listeners; 6) an audience of higher average occupational status; 7) an audience with more education; 8) an audience with a higher proportion of home owners; 9) unmatched listener loyalty. These factors would indicate that "good FM can be a highly satisfactory supplement for advertising specialized for the more expensive products".

The report discusses with refreshing honesty the reasons for FM's present plight; these need not be reviewed here. Suffice it to say that everyone concerned shares the blame to a certain extent. More important is the astute observation that "[apparently] all radio is headed for music, news, and specialized service programming, with most of the programs being musical. . . . The superiority of FM over AM on musical reception cannot be questioned".

Ray Stone of the Maxon agency conducted the survey and wrote the report. The project was undertaken for General Electric, whose high fidelity line Maxon handles. Since this is likely to be one of the most significant events in FM history, even if it is forgotten in the future, we'd like to offer both GE and Maxon our sincere congratulations.

BETWEEN the proof that a given process is technically possible, and its commercial realization, often lies a long and expensive period of experimentation, disappointment, development, and debugging. This was the case with video tape recording. It has long been known that it would eventually be possible to record TV programs on magnetic tape, and that — provided costs could be reduced — this would have great advantages over present "kinescope recordings" on film.

Several laboratories have been working on the problem for years. It is a genuine, brass-bound, copper-riveted problem, too: if a tape recorder of the conventional type were to be converted so as to record 4-Mc video, the tape speed would be something like 2,000 ips. A 14-inch reel of tape would last 29 seconds. Now, however, one company at least seems to have a solution; on April 14, Ampex Corporation demonstrated its new Video Tape Recorder. Tape width is 2 in., but operating speed is only 15 ips, which means that a full hour program can be preserved on a 14-inch reel. Horizontal resolution is said to be better than 320 lines; it is possible to transmit only 340. This is far better than a kinescope recording, and the gray scale is more accurately reproduced. On home sets VTR recordings should be indistinguishable from live pickups.

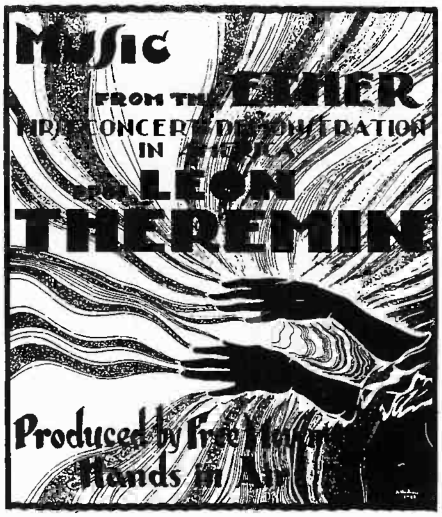
Such performance is achieved by using a new rotating-head system. The head assembly consists of four heads on a rotating drum. As one head leaves the tape, another makes contact with it. The recording is done in the same direction as tape travel, rather than perpendicular to it. And, although the tape itself travels slowly, the rotating head assembly gives an effective tape speed great enough to obtain a 4-Mc band width. Sound accompanying the picture information is recorded in the conventional way along one edge of the tape.

We understand that CBS Television has ordered the first three VTR units, and will use them to improve the quality of network broadcasts that are delayed regionally because of zone time differences. Although it is barely conceivable that recorders of this sort might become available for home use in recording TV off the air, they will certainly not be for a while yet. Present cost: \$75,000 each. — R.A.

MUSIC FROM ELECTRONS

by Robert Moog

ON the Tuesday evening of January 31, 1928, a capacity crowd filled the New York Metropolitan Opera House. The occasion was nothing as conventional as the performance of an opera, however. The audience braved the winter evening to witness what was then a remarkable novelty. Leon Theremin, a Russian physicist, was to perform on a musical instrument which he had developed and named after himself. The



METROPOLITAN OPERA HOUSE
Tuesday Evening, Jan. 31st, 1928 at 8:30 P. M.
Tickets $4.40 to $1.10 (Tax Included)
Now on Sale at Box Office
Wurlitzer Piano
RECITAL MANAGEMENT ARTHUR JUDSON
STEINWAY HALL NEW YORK

*Fig. 1. **Program cover for 1928 concert.***

instrument was made of radio components and was operated by electricity. Most important, however, was the fact that Professor Theremin would play his instrument *without touching it.*

The printed programs which the audience held set the proper mood: *Music from the Ether*, produced by a pair of hands reaching out from a raging inferno (see Fig. 1). Rudolf Wurlitzer introduced Professor Theremin and explained to the audience that Theremin would play his instrument by varying the position of his hands in the space around it. The Professor then proceeded to play. Although Theremin was a creditable musician, the audience was undoubtedly more impressed by the mysterious undulations of his hands than by the musical quality of his performance.

To show that the theremin was a worthy musical instrument as well as an intriguing novelty required the devoted efforts of several fine musicians. Fortunately, a small body of artists extended the frontier of theremin music so that all but the most conservative critics granted the theremin a place in the realm of "serious" music. The best known of these pioneers are Lucy Bigelow Rosen, Clara Rockmore, and Elena Moneak, who gave many public concerts in the 1930's, and Jeno von Takacs, who composed works for the theremin.

As a commercial venture, the manufacture of theremins was not a notable success. Shortly after Theremin's first concert the Radio Corporation of America bought his patent and began production of the instruments. Introduced at a time when few people were willing to invest in anything, the RCA theremins did not sell very well. After making only a few hundred instruments RCA discontinued production. With no instruments being produced, theremin music seemed to be on its way to oblivion. Only recently has there been a promising renewal of interest.

Two years ago we completed the design of a theremin which is played in the same manner as the RCA instrument, but which utilizes more modern circuits and components. In addition, it incorporates some features which were not present in the original. This instrument, the Model 351, will be described and compared to other musical instruments.

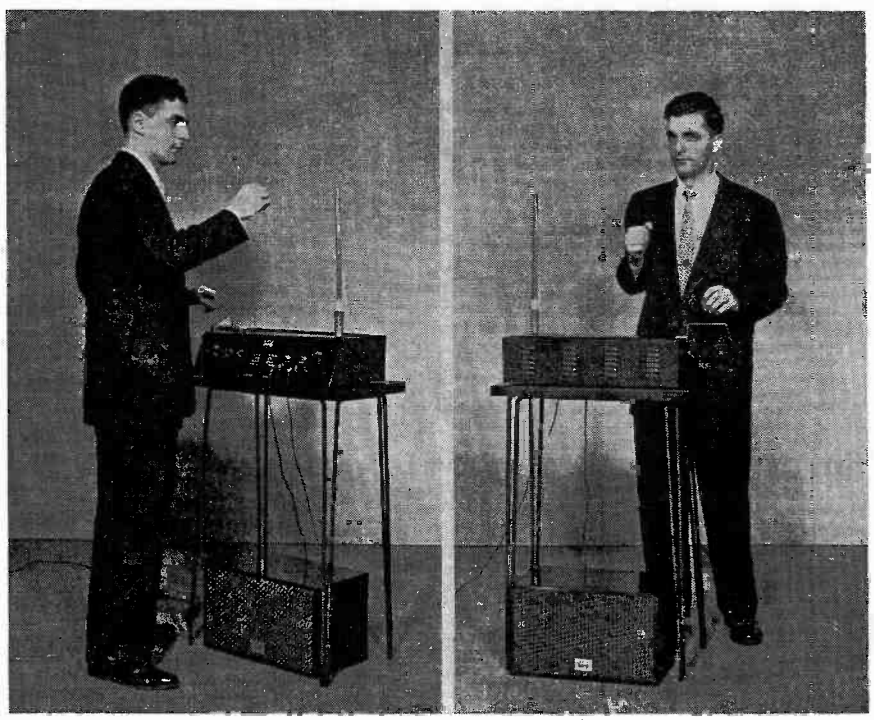
The Model 351 theremin is an electronic device played by movement of the performer's hands in the space surrounding it. Pitch of the tone is determined by the distance between the performer's right hand and the pitch antenna, Fig. 2, which is a long slender rod. Volume of the tone is determined by the distance between the performer's left hand and the volume antenna, a flat metal plate. Two switches on the front panel enable the performer to select the tone quality or timbre suitable to the music being performed. The entire instrument, except for the amplifier, is housed in a wooden cabinet 20 in. long, 11 in. deep, and 6 in. high. It weighs less than 20 lbs.

Pitch Control

Musical instruments are traditionally grouped in categories according to the way in which the tone is produced: string, brass, wood wind, and percussion. The theremin belongs in none of these categories, since its tone is produced by electronic circuits. This in itself is not enough to characterize it, however; we must also state that the pitch is controlled directly by the position of the performer's hands. If we were to divide musical instruments according to the way in which the pitch is controlled, we would have three categories:

Text continued on page 18

*Fig. 2. **Pitch, volume depend on the distance of player's hands from antennas.***



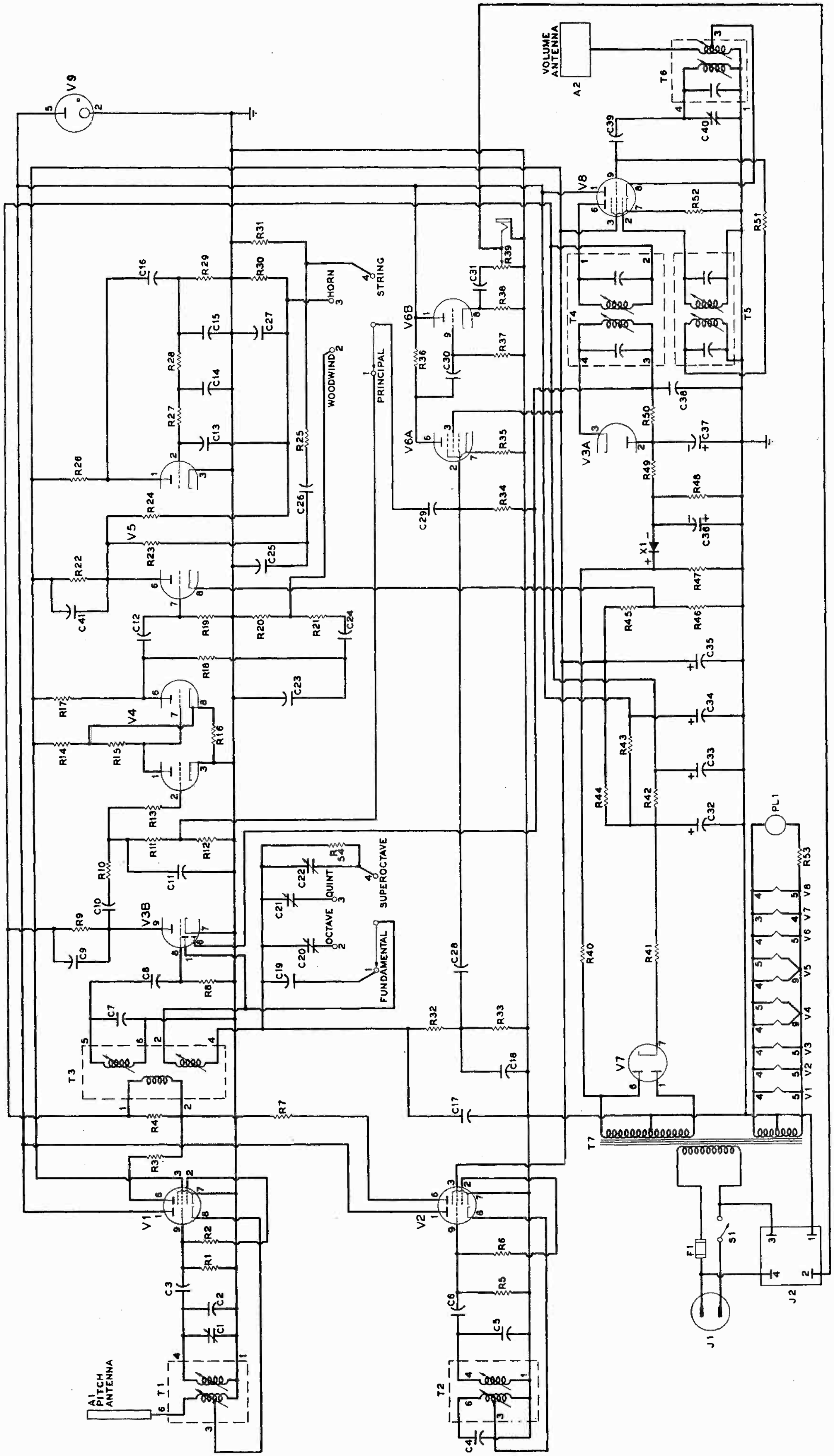


Fig. 3. Complete schematic of a commercial theremin. Outputs for a companion amplifier-speaker, and for a standard hi-fi system input, are supplied. Timbre and overtone are controllable.

Continued from page 16

1) Instruments which have a separate key or valve for every note (a piano or clarinet, for instance).

2) Instruments for which pitch is controlled partly by keys, and partly by the player himself (trumpet, French horn).

3) Instruments which have no keys, and for which pitch is controlled by the position of the performer's hands (stringed instruments, theremin). This list is arranged in order of increasing flexibility in pitch control. On the piano, only those pitches for which there are keys can be produced. With the theremin, however, any pitch can be produced. There are several reasons why this is desirable. First, it is often desired to go from one note to another in a glissando, or glide. Obviously, this cannot be done smoothly on a piano. Second, in many systems of harmony the intervals between notes are different from those of the traditional tempered scale. Even in playing classical music, departure from the tempered scale is often made. For instance, a good violinist may differentiate between G-flat and F-sharp but, on the piano, which is tuned to the tempered scale, G-flat and F-sharp are the same note.

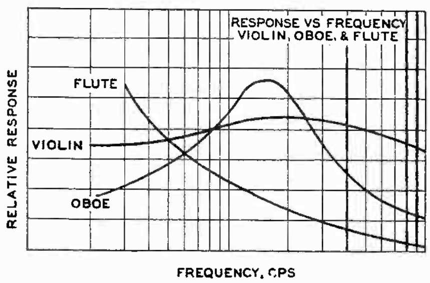


Fig. 4. Harmonic reinforcement curves for various musical instrument bodies.

So much for the musical aspect of pitch control for the theremin. How is this continuous pitch control achieved? The pitch circuit takes advantage of the fact that the hand is a conductor of electricity. Its connection to the rest of the body effectively grounds it. Therefore, the hand can be used as a grounded plate of a capacitor. If the hand is moved with relation to another electrical conductor, we have a variable capacitor. It is this variable hand capacitance which is used to control the pitch of the theremin.

Of course, the hand capacitance is very small — only a few micromicrofarads. It could not be used to tune an audio oscillator directly. A special type of pitch generator, called a beat frequency oscillator, is used in place of a conventional audio oscillator. The beat-frequency oscillator in the Model 351 theremin consists of two radio-frequency oscillators operating at frequencies very close together. The oscillator outputs are fed into a mixer circuit which effectively subtracts one frequency from the other. If the difference or beat frequency lies in the audio range, the mixer will deliver an audio output. For instance, if one RF oscillator is operating at 200 Kc and the other is operating at 199 Kc, the output of the mixer will be 1,000 cps. A small percentage change in frequency in one of the oscillators will result in a proportionally larger change in the audio output. Thus, with one of the RF oscillators operating at 200 Kc, the entire audio spectrum can be covered by changing the other oscillator frequency only 10%.

Fig. 3 is a schematic diagram of the Model 351 theremin. The beat-frequency oscillator which generates the pitch is composed of V1, V2, the triode section of V3, and their associated components. The oscillator coil (T1) of the variable oscillator V1 is designed to effect a relatively large change in the frequency of oscillation for a small change in capacitance of the pitch antenna caused by variation of hand capacitance. The fixed oscillator V2 is identical with the variable oscillator, except for the absence of a pitch antenna. The RF signals from the two oscillators are fed through mixing transformer T3 into a mixer, which is the triode section of V3. The output of V3 is passed through an RC filter composed of C9, R10, and C11, which removes the RF components and allows only the audio signal to pass.

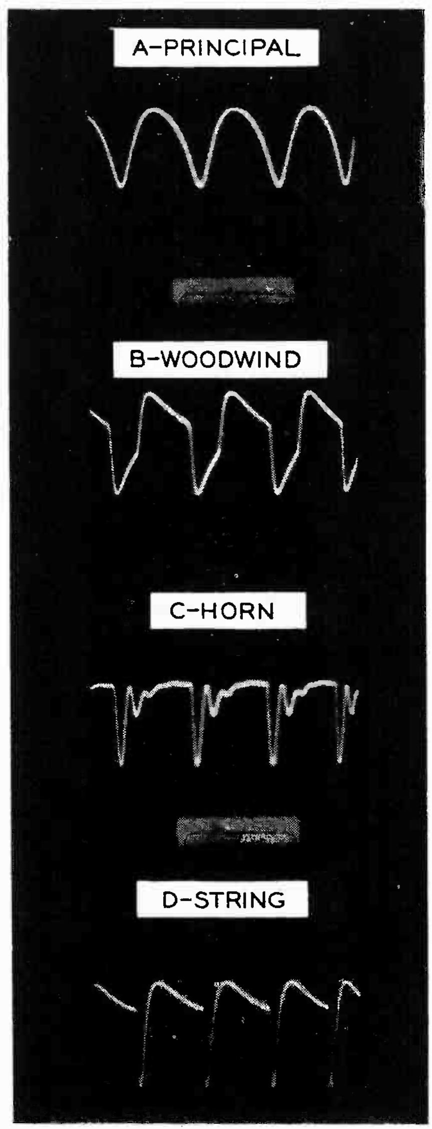
At that point the signal has very little harmonic content. This might appear at first to be desirable. Musicians often call a pleasing tone a "pure" tone, but it is usually far from pure in the sense of being free from overtones. For instance, the fundamental component of a violin tone is only a small part of the total. The remainder of the tone consists of harmonics, or overtones, whose frequencies are integral multiples of the fundamental frequency. These harmonics are not perceived by the listener as discrete tones, but instead give the fundamental tone an ear-pleasing timbre.

Harmonic Generation

In all conventional musical instruments, the tone source, be it a string, a reed, or the player's lips, generates all the harmonics of the fundamental tone. Before the sound is released into the air, it is transmitted through the body of the instrument, which attenuates some harmonics and allows others to pass. Thus it is the body of the instrument which determines, for the most part, its timbre. In the flute, harmonics are attenuated sharply, giving the tone a mellow quality. All harmonics are allowed to pass in the violin, although some are attenuated slightly. That is why the violin has a rich pleasing tone. The oboe body is highly resonant, reinforcing a narrow band of harmonics and attenuating the rest. The result is a sharp, nasal quality which makes the oboe tone easy to identify. Instrumental body transmission characteristics are shown in Fig. 4.

In the Model 351, the audio signal is passed through special circuits which introduce the desired harmonics, and then through attenuating filters which are electrical analogs of the body transmission responses of conventional musical instruments. The signal, as it enters V4, is shown in Fig. 5A. V4 and its associated circuitry act as a clipper, so that the signal emerges as a square wave. A square wave contains only odd harmonics, however. If you have ever heard a square-wave test on an amplifier, you know that a square-wave tone is hollow and woody, like that of a clarinet. While the tone may be pleasing, it is too distinctive to be the basis for more than one timbre. The next circuit, consisting of the left section of V5 and its associated components, forms a wide pulse from the square-wave input. This pulse, which contains the fundamental tone and *all* its harmonics, is fed into two filters. One is an RC filter which gives the signal a string-like quality. The other is a resonant filter involving a

Fig. 5. Outputs of tone shaping circuits.



phase-shift amplifier which gives the signal a sharp, horn-like quality. The outputs of these two filters, together with the output of a filter fed by square waves, and a signal taken directly from the beat-frequency oscillator, are all connected to switch S3. The performer can select the timbre which he desires simply by setting this switch. Wave forms of the four timbres are shown in Figs. 5A through 5D.

In addition to being able to select one of four timbres, the performer can also select one of three overtones. Note that the mixing transformer T3 has two secondaries. The upper secondary is broadly tuned to the fundamental frequency of the RF oscillators, and feeds the triode section of V3. The lower secondary is tuned by one of four capacitors which can be selected by switch S2. These capacitors are adjusted so that they tune the lower secondary to one of the harmonics of the RF oscillators. The lower secondary feeds its own mixing circuit composed of a diode (pin 1 of V3) and the RC filter R32, R33, C17, and C18. The output of this mixing circuit is the audio harmonic corresponding to the RF harmonic to which the lower secondary of the mixing transformer has been tuned. For instance, if the fixed RF oscillator is operating at 200 Kc and the variable RF oscillator is operating at 199 Kc, the output of the triode section of V3 will be 1 Kc. If the lower secondary of T3 is tuned (by one of the condensers connected to S2) to a frequency of 600 Kc, it will transmit the third harmonics of the fixed and variable RF oscillators, which are 600 and 597 Kc respectively. When these are mixed, the resultant will be 3 Kc, which is the third harmonic of the 1-Kc audio fundamental. With S2 the performer can select either the fundamental, second harmonic (octave), third harmonic (quint), or fourth harmonic (superoctave). The wave forms of these harmonics, when combined with the fundamental, are shown in Figs. 6A through 6D.

Once the proper harmonics have been added, the signal can be amplified and fed into a loudspeaker. The Model 351 contains one stage of amplification, which is part of the volume-control circuit.

Volume Control

Hand capacitance is used to control the volume also. A variable RF oscillator

Continued on page 33

Parts list for the theremin is shown at right. All parts, with the exception of the variable capacitors, antennas, and transformers, are stock items which are available through distributors of electronic parts. For further information on non-standard parts, write to the R. A. Moog Company, 51-09 Parsons Boulevard, Flushing 55, New York.

Part Number	Name and Type of Part	Value, Rating, Code
A1	pitch antenna	7-301
A2	volume antenna	6-306
C1, C40	variable capacitor	25 μμfd
C2, C5, C11	mica capacitor	560 μμfd
C3, C6, C17, C18, C39	mica capacitor	180 μμfd
C4	ceramic capacitor	15 μμfd
C7	mica capacitor	1,600 μμfd
C8, C9, C12, C23, C25	paper capacitor	.001 μfd, 600 volts
C10, C14, C15, C24, C26, C28	paper capacitor	.005 μfd, 600 volts
C29, C30, C13, C38	paper capacitor	.05 μfd, 600 volts
C16, C27, C41	paper capacitor	.01 μfd, 600 volts
C19	mica capacitor	2,200 μμfd
C20, C21, C22	trimmer capacitor assembly	2-251
C31	paper capacitor	0.22 μfd, 200 volts
C32	electrolytic capacitor	80 μfd, 400 volts
C33	electrolytic capacitor	40 μfd, 400 volts
C34, C35	electrolytic capacitor	20 μfd, 150 volts
C36, C37	electrolytic capacitor	10 μfd, 50 volts
F1	fuse	2 amp
J1	power connector	
J2	connector for model 400 amplifier	
J3	connector for amplifier other than model 400	
PL 1	pilot bulb	6.3 volts at 0.15 amp (No. 47)
R1, R2, R3, R5, R6, R7, R28, R36, R49, R51	carbon resistor	33 K, 1 watt
R4, R52	carbon resistor	2.7 K, 1 watt
R8, R9, R12, R16, R19, R21, R24, R25, R26, R33, R40, R47, R50	carbon resistor	220 K, 1 watt
R10, R14, R17, R18, R20, R22, R23, R27, R30, R32	carbon resistor	100 K, 1 watt
R11, R13, R34, R37	carbon resistor	1 MΩ, 1 watt
R 15, R54	carbon resistor	10 K, 1 watt
R29	carbon resistor	10 MΩ, 1 watt
R31, R48	carbon resistor	56 K, 1 watt
R35	carbon resistor	1.8 K, 1 watt
R38	carbon resistor	5.6 K, 1 watt
R39	volume control potentiometer	10 K
R41	wire wound resistor	1 K, 10 watts
R42	wire wound resistor	1 K, 5 watts
R43	wire wound resistor	5 K, 5 watts
R44, R45	wire wound resistor	7.5 K, 5 watts
R46	carbon resistor	150 ohms, 1 watt
R53	carbon resistor	22 ohms, ½ watt
S1	power switch (this is part of a silencing switch on later models)	
S2, S3	lever switch	4 position, 1 pole
T1, T2	pitch oscillator transformer	No. 3-301
T3	coupling transformer	No. 3-303
T4, T5	bandpass transformer	No. 3-306
T6	volume oscillator transformer	No. 3-302
T7	power transformer	No. 4-301
V1, V2, V6, V8	vacuum tube	6U8
V3	vacuum tube	6T8
V4, V5	vacuum tube	12AX7
V7	vacuum tube	6X4
V9	regulator tube	OA3
X1	selenium rectifier	130 volts at 20 ma

by E. B. Mullings

Using Test Instruments

Audio Signal Generators, Part I

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

THE first three installments of "Using Test Instruments" were devoted to the vacuum-tube voltmeter. Information was given on the theory of operation, the mechanics of employing such an instrument, and some of the tests that could be made with the VTVM.

Perhaps the next most logical item of test equipment to turn our attention to would be the *audio signal generator.* If the VTVM qualifies as the first item of test equipment a beginner should obtain, the audio generator would very likely rate second in importance. Used in conjunction with a VTVM, the audio signal generator makes it possible to run frequency-response tests on high fidelity equipment, and it functions as a signal source for many other tests and observations.

An audio signal generator is, as the name implies, a source of audio-frequency sine-wave signals, to be used wherever such a "known" signal is required. The signal frequency is usually variable; the amount of audio output is generally controllable and is often metered. A typical modern audio signal generator is shown in Fig. 1.

An audio generator consists, basically, of an oscillator circuit capable of creating a sine-wave signal within the audio range. Perhaps our discussion should begin with some explanation of how a simple oscillator functions.

A typical oscillator circuit employs a vacuum tube and is similar to an amplifier circuit, except that a path is provided for feedback from the plate circuit to the control-grid circuit. Oscillation occurs because this feedback path is adjusted so that voltage from the plate circuit is fed back *in phase* with the grid voltage. This reinforces the tendency of the grid to swing from cut-off in the negative direction to saturation in the positive direction, so that sustained oscillation is produced. The control grid swings first positive and then negative, creating a signal. This signal appears in the plate circuit of the stage, and is fed back to the control grid to reinforce the "swinging" action.

A very simple oscillator circuit is shown in Fig. 2. An audio transformer is connected so that the control grid and the plate of a vacuum tube are coupled together. It is important to note that the voltage fed back from the plate must be in phase with the grid voltage. Normally, the plate voltage is 180° out of phase with the grid voltage. That is, when the control grid of a tube becomes more positive, the plate becomes more negative, and vice versa. If the plate signal were fed directly back to the grid, it would be 180° out of phase and tend to cancel the grid signal; degeneration, not oscillation, would occur. Since there is a 180° phase shift between the primary and secondary windings of an audio transformer, however, the plate signal is brought back into phase with the grid signal by virtue of the transformer used between plate and grid, Fig. 2. This circuit would oscillate, because there would be regeneration in the stage. It is evident that regeneration is essential to oscillation, and that some means must be provided to bring the signal fed back from the plate into phase with the voltage at the grid. Regeneration requires positive (in-phase) feedback to the grid, and degeneration requires negative (out-of-phase) feedback.

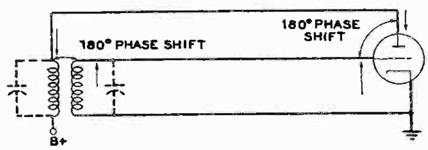
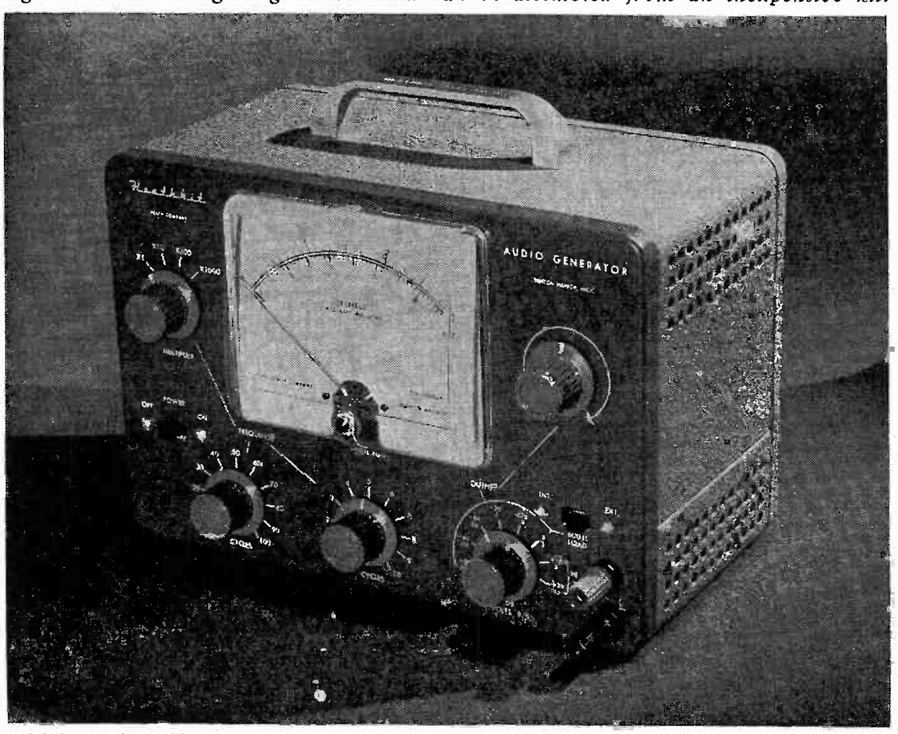


Fig. 2. Basic circuit for an oscillator.

It may be helpful to examine the operation of the circuit in Fig. 2 more closely. Let us say that a small power-line fluctuation causes the voltage on the plate of the tube to drop slightly. This voltage drop is coupled through the transformer and reversed in phase, so that the control grid is made slightly positive. That, in turn, increases the plate current and reduces the plate volt-

Fig. 1. An audio signal generator that can be assembled from an inexpensive kit.



COURTESY HEATH COMPANY

age still more; the drop is again coupled through the transformer as a positive swing, and so on. Eventually the point is reached when no greater tube current can flow. Then, because no further change in voltage is coupled to the grid, it cannot remain positive. As soon as it becomes less positive, plate current decreases and plate voltage begins to rise; this is coupled as a negative voltage

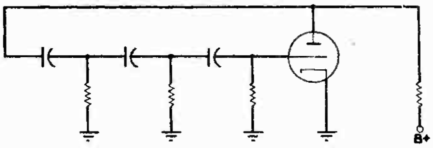


Fig. 3. Phase shift by an RC network.

to the grid, which further decreases plate current. When the grid approaches that voltage at which plate current is cut off, the voltage coupled through the transformer again decreases, because the rate of change in current is decreasing. Again the grid begins to go positive, and continues to do so until saturation is reached. This process repeats again and again, in the continuous process of oscillation.

Such a circuit will oscillate very satisfactorily at a single frequency. The frequency would be determined by the characteristics of the transformer used, and by the stray capacitance existing across transformer windings. Connecting various capacitor values across the primary or secondary windings of the transformer would change the oscillator frequency within a limited range.

The circuit in Fig. 2 would not be practical as an audio generator for several reasons. Limited frequency range, the physical size of the transformer that would be required, and other factors would inhibit its use.

There are many other types of basic audio oscillators, one of which is illustrated in Fig. 3. This is an RC oscillator circuit, in which the time constant of a series of resistors and capacitors in the grid circuit is used to accomplish the essential 180° phase shift, bringing the plate signal back into phase with the grid signal so that regeneration and sustained oscillation can occur. Changing the value of resistance or capacitance will change the frequency of oscillation for the circuit. This circuit comes closer to that used in a modern audio generator, and offers advantages over the transformer-coupled circuit in Fig. 2. But for extremely low distortion of the output sine wave, combined with flexibility, it still leaves something to be desired.

The circuit of the Heathkit Model AG-9 Audio Signal Generator, used as an example, can best be explained by starting with a block diagram of the oscillator section. This is given in Fig. 4.

This oscillator circuit employs a 6AU6 tube and a 6CL6 tube. The 6AU6 functions as a pentode voltage amplifier, while the 6CL6 functions as a triode-connected cathode follower. Both *degenerative* and *regenerative* feedback is provided between these two tubes. Regenerative feedback, alone, would allow sustained oscillation between the two tubes, producing a signal. When degenerative feedback is employed, it nullifies the effect of the regeneration, and oscillation ceases. Oscillation can only occur under these circumstances if the degenerative feedback is reduced.

Fig. 5 extends the block diagram one step further by making the degenerative feedback "frequency selective". In other words, the circuit providing degeneration is designed in such a way that degeneration is at a maximum except at a certain frequency, preventing oscillation at any frequency but that particular one. With minimum de-

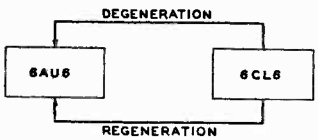


Fig. 4. Multiple feedback loop system.

generation at one frequency, the circuit will oscillate at that frequency. Using both positive and negative feedback tends to minimize distortion of the sine-wave output.

This is exactly what happens in the oscillator circuit of the Heathkit Model AG-9. Regeneration is relatively constant, while the normal amount of degeneration is sufficient to prevent oscillation except at a given frequency. When degeneration is reduced at a particular audio frequency, oscillation takes place at that frequency. Oscillation is, therefore, controlled by setting the frequency at which degeneration will be at a minimum. The result is an extremely stable yet flexible oscillator that will produce an output signal with very low distortion. The schematic diagram of the AG-9, Fig. 6, reveals how this is done.

Notice that the feedback signal is taken across the 5,000-ohm resistor in the cathode circuit of the 6CL6 cathode follower. It is applied through a 20-

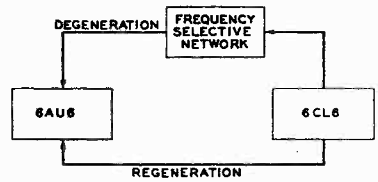
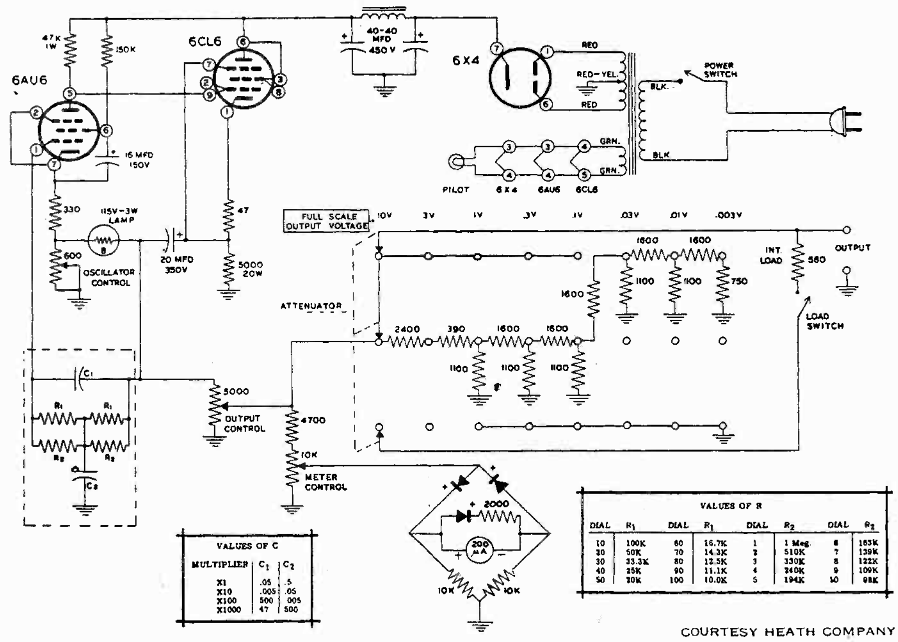


Fig. 5. Adjustable-frequency oscillator.

μfd capacitor and a 3-watt lamp bulb to the cathode of the 6AU6 stage, furnishing regeneration. The same signal is fed through a resistor-capacitor network (enclosed by a dotted line), and then up to the control grid of the 6AU6 for degeneration. By changing the values of the capacitors and resistors used in the RC network, the frequency for which degeneration will be at a minimum can be controlled. In this way, the frequency of oscillation is controlled. The RC "notch" network in the grid of the 6AU6 is really the frequency-determining portion of the oscillator circuit. Oscillation occurs at the notch frequency, where degeneration is minimum and phase shift is zero.

The 3-watt lamp in the regenerative feedback path maintains the amplitude of oscillation at a nearly constant value. Regeneration is applied through a voltage divider consisting of the lamp and the oscillator control. An increase in

Fig. 6. Complete circuit of a practical audio signal generator (shown in Fig. 1).



VALUES OF C		
MULTIPLIER	C_1	C_2
X1	.05	.5
X10	.005	.05
X100	500	005
X1000	47	500

VALUES OF R							
DIAL	R_1	DIAL	R_1	DIAL	R_2	DIAL	R_2
10	100K	60	16.7K	1	1 Meg.	6	163K
20	50K	70	14.3K	2	510K	7	139K
30	33.3K	80	12.5K	3	330K	8	122K
40	25K	90	11.1K	4	240K	9	109K
50	20K	100	10.0K	5	194K	10	98K

COURTESY HEATH COMPANY

output increases the lamp current, the lamp temperature, and the lamp resistance. This reduces the amount of feedback applied to the 6AU6 cathode, and the output is thereby reduced. A balanced condition is thus obtained. The oscillator control is used to set the nominal output level.

The notch network is a capacitor-shunted, bridged-T type, and consists basically of 2 resistors and 2 capacitors. The "notch" occurs at a frequency

$$f = \frac{1}{2\pi RC},$$

where $C = \sqrt{C1\,C2}$.

From the relationship shown, it is evident that a decrease in capacitance by a factor of 10 will increase the frequency by a factor of 10. As the values of C1 and C2 were chosen with a 10:1 ratio, 5 capacitors can do the job of 8 in achieving 4 decade frequency ranges. For control of the frequency within these steps of 10, the value of resistance is changed. Controlling resistance and capacitance in this fashion makes it possible to simplify the control of oscillator frequency, so that two significant figures and a multiplier may be selected on front panel selector switches to set the frequency of oscillation. Thus, changing resistance and capacitance in the notch network and, therefore, changing the frequency at which the notch occurs, can control the frequency of the audio generator from 10 cps to 100 Kc.

The attenuator circuit for the Model AG-9 functions to reduce the output to predetermined values. It provides output voltage settings of 10, 3, 1, 0.3, 0.1, .03, .01, and .003 volt maximum. It is located to the right of the oscillator section of the circuit, Fig. 6. Notice that the output from the oscillator is first applied to a continuously variable potentiometer labeled OUTPUT CONTROL, then across a voltage-divider network which feeds voltage both to the metering circuit and to the attenuator circuit. By means of switched resistors the attenuator circuit divides and then sub-divides the total voltage, taking a smaller and smaller percentage, and applies it to the output terminals.

Attenuator action can be seen more clearly by referring to Fig. 7. The voltage across R2 and R3 is first applied to the output terminals of the generator, then the voltage across R3 by itself. Then two resistors are connected in parallel with R3, further dividing the voltage appearing across R3, and the output voltage is tapped out so that the voltage across R5 is applied to the output. Two resistors are in parallel with R5, further dividing the output, and then the voltage is taken from across R7, and so on. Each successive step divides the output into a smaller percentage of the maximum output so the signal appearing at the output terminals of the generator may be reduced in predetermined steps. The circuit in Fig. 7 is essentially the same as that in the larger diagram.

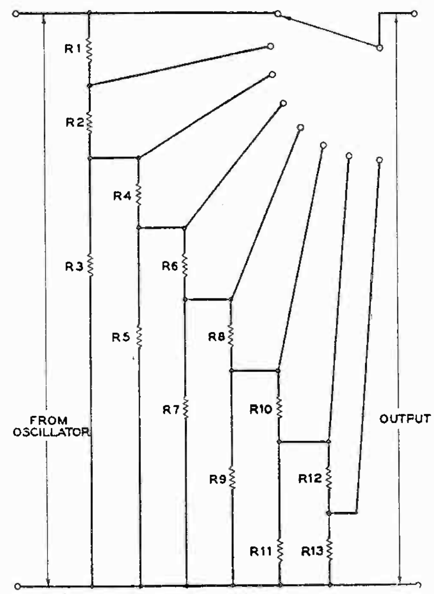


Fig. 7. The output attenuator section.

The attenuator system is designed for 600 ohms output up through 1 volt, and high-impedance output at the 3- and 10-volt positions. The 600-ohm positions can be terminated by an internal load for high-impedance work, or the internal load can be disconnected when an external 600-ohm load is used. In the 3- and 10-volt positions, of course, this internal load is disconnected automatically. The attenuator operates in steps of 10 db.

A large meter on the front panel of the Model AG-9 reads the output signal voltage. The schematic will reveal that the meter is connected to measure a proportion of the voltage appearing at the input to the attenuator circuit. However, the scales of the meter correspond to the voltages available with the attenuator, and the actual output may be read on the meter by employing whichever scale is appropriate to the attenuator position being used. One scale reads 0 to 3, and the other 0 to 10. In addition, a db scale is provided on the meter.

Continued on page 32

The Village Soundsmith*

Under a purple neon sign
The hi-fi salon stands,
Where sold are speaker cabinets
Whose walls are filled with sands,
And radio-detector kits
With FM-AM bands.

The window is a grand display
Of shiny amplifiers —
Their bottom plates removed to show
A maze of colored wires —
And pickup arms and tape machines
To lure prospective buyers.

Within, the shop is well designed
To please the audiophile,
With carpeting upon the floor
Of deep and heavy pile,
And ceiling covered, wall to wall,
With sound-absorbent tile.

Day in, day out, from nine to six,
You hear the systems playing,
While patrons listen, mute and grim,
The sundry merits weighing.
And sound waves bounce around the room,
Eventually decaying.

The music lovers gather here,
Come early and stay long.
They like to hear as brass and strings
Their voices blend in song;
They thrill to roll of kettledrum
And shimmer of the gong.

The children coming home from school
Are drawn into the store.
They like to watch the tubes light up
And hear the woofers roar,
And feel the organ pedal notes
That agitate the floor.

The dealer, though he labors long,
Takes time, whenever able,
To show you how to check your speeds
And oil your old turntable,
Or help the kid who came to buy
A length of shielded cable.

The shop is ever filled with folk
Who come from miles around
To see the shop and taste the wares
And hear the man expound.
Did blacksmith ever draw such crowds
As he who deals in sound?

K. A. ALEXANDER

***With a deferential nod to H. W. Longfellow.**

why equalize by herman burstein

BY means of tone controls, loudness controls, and sharp-cutoff filters in high fidelity systems, the user can alter tonal balance to compensate for room acoustics, program source defects, and personal preferences. Thus listening pleasure may be enhanced by departures from flat response. At the same time, a prime requisite of a high fidelity system is that, at well-defined settings of such controls, it should deliver to the speaker an electrical response flat over a range of at least 30 to 15,000 cps. Flat response is a reference point to which the listener may always return when compensating for different program sources or different listening conditions.

Although flat response after equalization is a basic requirement, most of the program sources commonly handled by a high fidelity system are not initially balanced at all frequencies. FM radio (including TV sound), discs, and pre-recorded tapes are unbalanced in the bass and/or treble range. Only AM radio, within its limited range of 5,000 cps or occasionally 10,000 cps, comes in flat. Obviously, then, an important function of high fidelity components is to provide accurate equalization which restores flat response.

The audiophile may justifiably ask, if flat response is required at the amplifier output, why not start with flat response at the sources? The answer, in broad terms, is that uniform frequency response is not the only requirement of high fidelity. Of at least equal importance are low noise and low distortion. It has been found that the best compromise among all three requirements can be reached by shaping the response curve of the program source and then reshaping it in the reproducing system.

The following discussion is intended to give a more detailed answer to that question, with specific reference to FM, discs, and pre-recorded tape. This knowledge should be desirable not only for its own sake, but also as a means of obtaining a better understanding of the high fidelity system so that it can be used to greater advantage and sources of malfunction recognized more easily.

FM Equalization

The Federal Communications Commission requires FM stations to provide a specified amount of treble boost or pre-emphasis, as shown in the upper curve, Fig. 1. An FM tuner must provide a complementary treble droop, or de-emphasis, which offsets the pre-emphasis so as to produce flat response. Obviously, an FM tuner with treble de-emphasis dissimilar to the lower curve does not provide flat response.

The reason for treble pre-emphasis at the FM transmitter involves two factors: 1) the relatively low amount of audio energy at treble frequencies; and 2) the relatively large amount of noise at treble frequencies. The objective is to maximize the signal-to-noise ratio — that is, the amount of desired audio energy to undesired noise energy. Through treble boost at the FM station, the ratio between audio energy in the treble range, and noise produced in the transmitter, is increased. At the receiving end, FM tuner noise is cut down at the same time that treble response is reduced to normal.

Investigations have shown that typical musical sources and the human voice produce their highest average energy in the vicinity of 400 cps. Using 400 cps as a reference point, average signal level at 10,000 cps may be down 13 db or more; in other words, the average signal voltage produced at 10,000 cps may be less than one-fourth the average signal voltage at 400 cps. It is no coincidence (Fig. 1) that at 10,000 cps the amount of treble boost required by the FCC is of about the same order as the decline in audio energy. These two factors compensate each other, so that the average audio modulation *actually transmitted* is about the same at all frequencies.

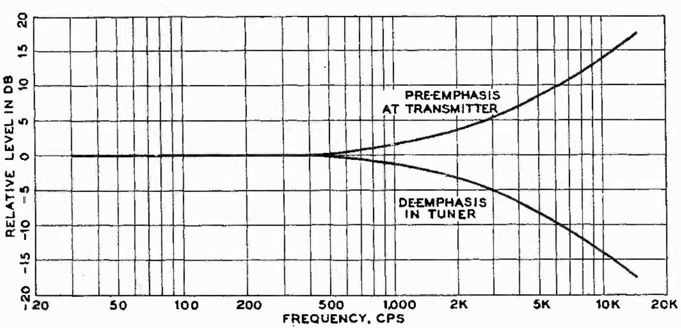
Audio energy declines below 400 cps as well as above this point. Why, then, doesn't the FCC also require bass boost at the transmitter and bass cut in the receiver so as to increase the signal-to-noise ratio? Because noise predominates in the upper portion of the audio range.

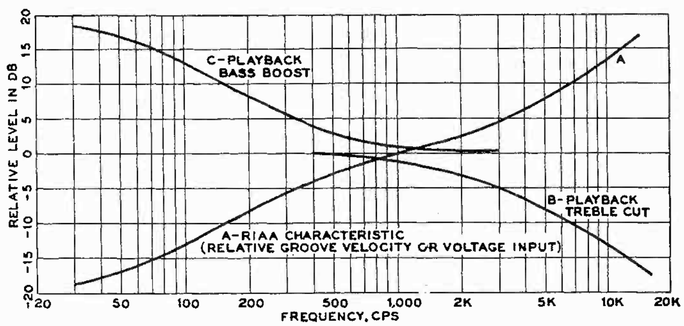
Noise consists of random energies more or less equally distributed per cps throughout the spectrum. That is, all discrete frequencies have equal chances of being represented in the noise complex. Since the treble range contains a greater number of individual frequencies than the bass range (there are 16 times as many frequencies in the 4 octaves from 800 to 12,800 cps as in the 4 octaves from 50 to 800 cps), noise appears to have essentially a high-frequency characteristic.

Disc Record Equalization

Most disc recordings now made in the United States, and many of those made abroad, have equalization conforming to the RIAA curve. This is the standard established by the Record Industry Association of America. Discs contain both high-frequency boost and low-frequency cut. Therefore, the playback system

Fig. 1, left: treble pre-emphasis in FM transmitter, and complementary de-emphasis in tuner. Fig. 2, right: disc record curves.





must provide a complementary amount of high-frequency droop and bass boost. Curve A in Fig. 2 shows the equalization used in recording, while Curves B and C show the equalization required in playback when a magnetic pickup is used.

Some of the records presently manufactured, and a great number of those made two years ago or earlier, require equalization somewhat different than the RIAA standard. Such curves — among them the LP and Old AES characteristics — differ from the RIAA curve by appreciable but not very great amounts. These departures from the standard can be quite satisfactorily corrected by the use of bass and treble controls, if desired, although most high fidelity systems provide specific controls for matching a variety of equalization curves.

For convenience, the following discussion is in terms of the RIAA standard, although it applies in principle to the other curves as well.

Why does the RIAA recording curve have treble boost, as shown in Fig. 2? The reason is very similar to that for using treble boost in FM. Much of the noise developed in the use of discs occurs in playback, being due to minute irregularities, bits of dust, etc. in the record grooves. Because audio energies are relatively small at the upper frequencies, the amount of desired audio information compared with undesired noise would be unsatisfactorily low. Thanks to treble pre-emphasis, however, playback noise is reduced when treble cut is used to restore frequency balance.

In deciding how much treble pre-emphasis should be used in recording, the RIAA had to consider that excessive treble boost would cause technical problems. The higher the recorded level at any given frequency, the greater is the speed at which the cutting stylus travels; and the greater is the speed of the playback stylus, which follows the path explored by the cutter. But excessive speed of the playback stylus results in distortion. Accordingly the RIAA had to settle for about the same recording speed at all treble frequencies. But to the extent that average audio energy declines at high frequencies, it is possible to use a corresponding amount of treble boost; the net result is more or less constant speed of the playback stylus over the treble range on average recorded material.

The RIAA curve assumes the same distribution of audio energy as does the FM pre-emphasis characteristic. Therefore the treble portion of Curve A in Fig. 2 (treble boost in recording) is essentially the same as the pre-emphasis in Fig. 1 (treble boost in FM transmission).

Now let us consider why the RIAA characteristic requires bass cut in recording, with complementary boost in playback when a magnetic pickup is used. Here it is necessary to introduce two everyday words, velocity and amplitude, in their technical meaning and relationship to each other.

At a given frequency the cutting stylus moves from side to side, chiseling out of the rotating disc a groove having the shape of the audio wave form. In the absence of equalization, the inherent nature of the cutter is such that it would travel at constant velocity — that is, speed — at all frequencies if input voltage were kept the same. At low frequencies, then, the cutter must make greater sidewise excursions than at high frequencies in order to maintain the same velocity; in other words, the groove has greater amplitude. As a parallel case visualize two inebriates, each capable of walking 3 miles per hour. The first one goes down a wide street and staggers from curb to curb, while the second goes down a narrow street and does the same. The sidewise alterations of the first drunkard are of large amplitude and therefore of low frequency; those of the second are of small amplitude (because the street is narrow) and therefore of greater frequency.

In short, as frequency goes down, the groove amplitude must increase with constant input voltage if the velocity is to remain the same. This is illustrated in Fig. 3.

If the cutting stylus maintained constant velocity throughout the bass range, the amplitude of the groove would become so great that, taking into account the normal spacing between grooves on a commercial record, one groove would overlap another. The result would be distortion.

Thus it is necessary to reduce the amplitude of the cutting stylus excursions in the bass range. This is done through bass cut, as shown in the lower portion of Curve A, Fig. 2.

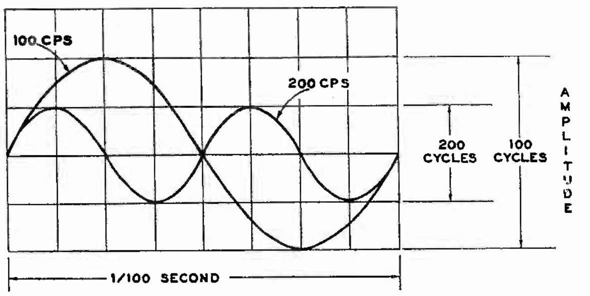
In determining the amount of bass droop that should be used in recording, the RIAA had to consider the playback problems encountered by a magnetic cartridge, which requires a corresponding amount of bass boost. Hum picked up by the cartridge and hum generated in the preamplifier are accentuated to the extent of the required bass boost in playback. Also, rumble is amplified by the same amount. The RIAA could go only so far, then, in requiring bass cut in recording. The limit decided upon gave consideration to the fortunate fact that average audio energy declines below 400 cps, which partially offsets the increase in amplitude as frequency drops. The RIAA curve prescribes enough additional bass cut to insure that below 500 cps the recorded amplitude stays more or less constant for constant input voltage, thereby preventing overcutting the grooves.

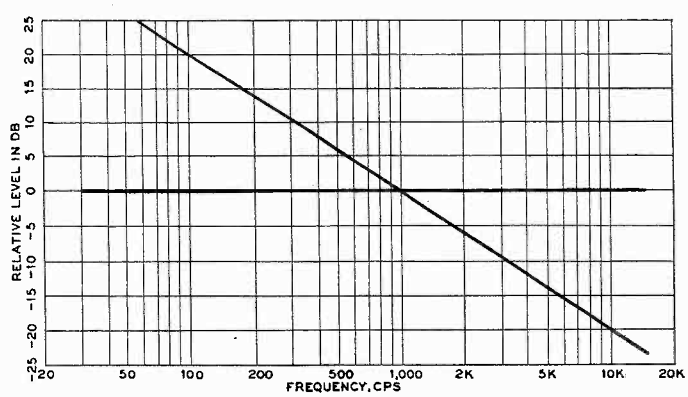
It has been indicated several times that this discussion relates to magnetic pickups, which are the type most commonly employed for high fidelity purposes. It bears repetition and emphasis that Fig. 2 and the associated text relate only to magnetic cartridges.

The magnetic pickup is a velocity device, the same as a cutter. It puts out the same voltage at all frequencies so long as its velocity remains the same. Thus it is the electrical counterpart of the cutting stylus, which cuts at the same velocity at all frequencies if input voltage is held constant. The curves in Fig. 2 may therefore be viewed as voltage curves or velocity curves, because they both represent devices for which voltage and velocity are directly proportional to each other.

Fig. 2, however, is not appropriate to piezo-electric (ceramic and crystal) and capacitive pickups. These are so-called

Fig. 3, below, and Fig. 4, right: amplitude must increase with decreasing frequency for velocity to remain constant.





amplitude devices and require completely different playback equalization, as discussed in the next section.

Amplitude Pickups

If input voltage is held constant as frequency changes, the amplitude of the recorded groove decreases as frequency rises, while recorded velocity stays the same. Fig. 4 shows the relationship between a constant-velocity recording (constant signal input) and the resulting groove amplitude as frequency varies. Curve B, amplitude, drops at a rate inversely proportional to frequency. When frequency doubles, Curve B falls to half its former level; translated into db, half the level represents a decrease of 6 db. Thus Curve B is known as a 6-db-per-octave curve.

Since Curve A in Fig. 4 is not the actual RIAA characteristic used in recording, neither is Curve B the actual amplitude characteristic. The velocity characteristic is that of Curve A in Fig. 2. In order to translate the latter into amplitude, it is merely necessary to adjust Curve A in Fig. 2 for the difference between velocity and amplitude at each frequency that is shown in Fig. 4. The net result is Curve A, Fig. 5. This is the RIAA recording characteristic expressed in terms of amplitude.

It is an inherent characteristic of piezo-electric and capacitive pickups to produce a voltage proportional to lateral displacement of the stylus, *not* stylus velocity.

The amount of displacement is determined by groove amplitude. Consequently, these pickups are amplitude-responsive devices. Curves B and C in Fig. 5 show the bass cut and treble boost required in playback when using an amplitude pickup. This situation is just opposite to that for a magnetic cartridge, which requires bass *boost* and treble *cut*.

Most preamplifiers, however, do not contain specific equalization for amplitude pickups. Instead, the situation is handled in one of two ways:

1) The pickup is loaded with (connected across) a resistor or more elaborate network which, in conjunction with the electrical characteristics of the pickup, converts the cartridge into a velocity device — one that turns out a voltage proportional to velocity rather than amplitude of stylus movement. Then the pickup may be treated just as a magnetic cartridge, connected to a preamplifier input designated for a magnetic cartridge and equalized accordingly. Most manufacturers of amplitude pickups provide adapters for converting their cartridges into velocity devices.

2) The amplitude pickup is fed into a standard unequalized input circuit, such as that intended for a tuner, and the equalization requirements are met as follows: bass cut is achieved by using a load resistor of proper value. Treble boost is incorporated in the cartridge itself, which has a rising characteristic in the treble range as the result of carefully controlled resonance.

Tape Recorders

Since equalization requirements vary with tape-recorder speed (as well as playback-head gap width, bias, and tape) it will be assumed that a speed of 7.5 ips is used. This speed is fairly standard for high fidelity home use, offering a satisfactory compromise between tape economy and extended high-frequency response. Most commercial pre-recorded tapes operate at 7.5 ips.

Although there is yet no standard equalization for 7.5 ips, it has become common practice at this speed to use the standard established for 15 ips by the National Association of Radio and Television Broadcasters. The signal on a tape conforming to the NARTB characteristic has a slight amount of bass boost and a pronounced amount of treble droop, as shown in Fig. 6.

It should not be thought for a moment that the treble decline shown in Fig. 6 results from purposeful treble cut in the recording amplifier of a tape machine. Quite the contrary: the recording amplifier introduces a very substantial amount of treble boost. In the process of impressing a signal on the tape, however, several phenomena occur, including demagnetization and bias erase. These cause recording losses that increase in severity as frequency rises.

The losses can be overcome by treble pre-emphasis in recording. But too much signal applied to the tape causes distortion from overloading. To avoid overloading, roughly the same amount of signal should be delivered to the tape at all frequencies. Since average audio energy declines in the upper range, the treble frequencies can be boosted *up to a point* without causing overload. At 7.5 ips, the permissible amount of treble boost (based on the variation of average audio energy with frequency) is appreciably less than the amount of recording losses. The net result is that the recorded signal on the tape exhibits a high-frequency droop. The NARTB standard stipulates the amount of this droop.

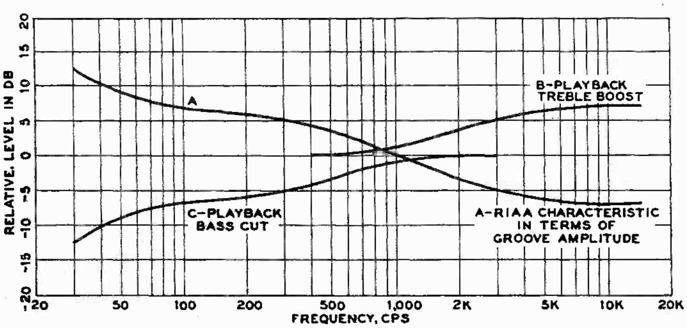
Before turning to the equalization requirements in playback, which are *not* the converse of Fig. 6, it is necessary to examine the characteristics of the playback head. Like a magnetic phonograph cartridge, the playback head is a velocity device. Increasing frequency means that the magnetic field on the tape is changing from instant to instant at a faster rate; it has a higher velocity. Therefore, given a tape bearing signals of equal recorded strength at all frequencies, the playback head produces an output rising with and proportional to frequency.

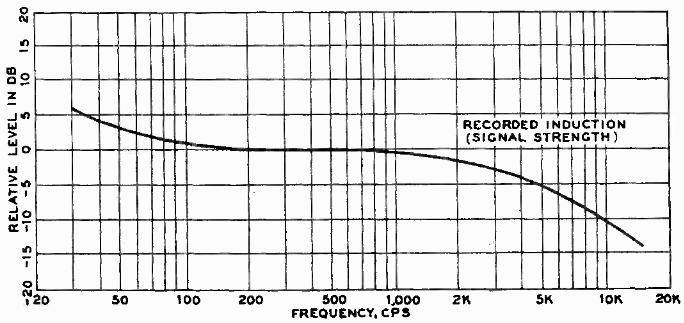
Response of a perfect playback head is shown by the solid line in Fig. 7. Actually, at 7.5 ips, there are losses in the extreme treble range which do not begin to be appreciable until about 13,000 cps in the case of a high-quality head. These losses may be ignored here since a tape recorder that gets out to 13,000 cps at 7.5 ips, yet maintains low distortion and a high signal-to-noise ratio, is doing very well. If the NARTB requirement of response no more than 4 db down at 15,000 cps were to be met, a slight amount of playback treble boost would be needed.

As the result of the playback head's rising response with frequency, it is necessary to offset the curve in Fig. 7 by playback equalization that rises as frequency declines. This equalization should not cover the same range as in Fig. 7. It will be remembered, referring in Fig. 6, that a tape recorded with the NARTB characteristic has treble droop. Therefore, part of the playback head's rising characteristic is used to offset the recorded droop. This is shown in Fig. 8, where the portion CB of the play-

Continued on page 37

Fig. 5, left: RIAA equalization for an amplitude phono pickup. Fig. 6, right: tape-recorded signal with NARTB equalization.





An AUDIOCRAFT *kit report*

Heath WA-P2 Preamp-Control Unit

AFTER opening the package containing parts for the WA-P2 preamplifier-equalizer-control unit kit, and checking them off against the parts list, the first question that occurs to everyone is "How can they do it for $20?".

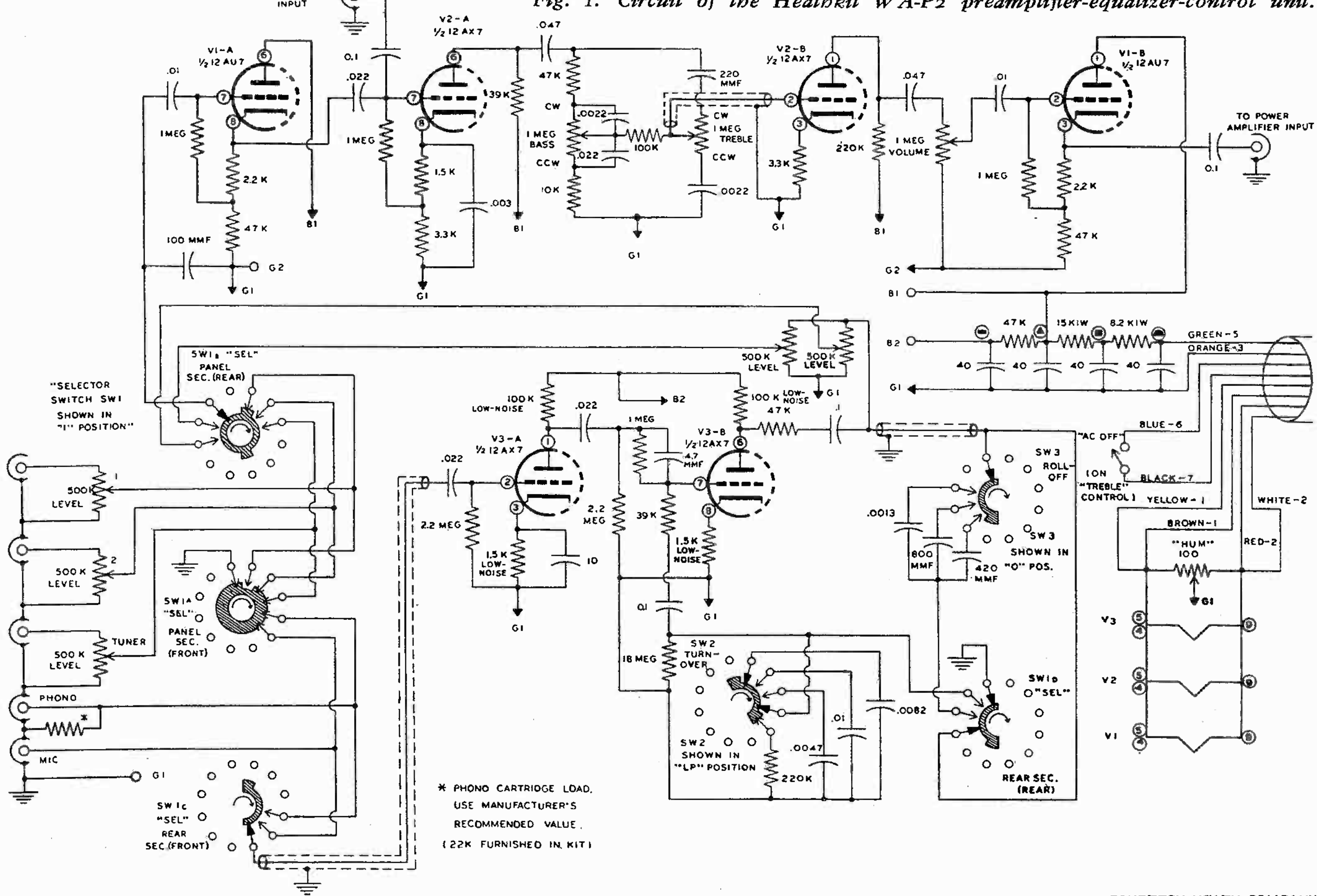
It does seem impossible. This kit utilizes three dual-section tubes in an elaborate and fairly complex circuit; there are individual input channels for a magnetic pickup cartridge, a microphone, and three high-level sources, with input level controls for all of them. There are individual phono turnover and rolloff controls, separate bass and treble tone controls, a selector switch, master volume control, and two low-impedance output circuits. Furnished with the kit are 31 resistors, 9 continuously variable controls, 4 switches, 26 capacitors, 3 tubes, 40 major pieces of hardware and 156 minor pieces, and a substantial amount of wire, cable, spaghetti, and shielding. The net cost of this and a 36-page step-by-step instruction book is $19.75. Properly used, too, this preamp can deliver performance equal to that of most of the highly respected (and expensive) units. Its appearance is good. It is a genuine bargain, no matter how you look at it.

Circuit Layout

Fig. 1 is a schematic diagram of the WA-P2. The five input jacks at the left are located on the back panel, as shown in Fig. 2, with a level control for each input circuit directly above the corresponding jack. High-level inputs, marked 1, 2, and TUNER, are connected directly to their level controls and then to the selector switch. One section on this switch grounds all inputs except the one selected, so as to minimize background chatter on the active channel from the other sources.

The PHONO and MIC inputs go to separate contacts on another section of the selector switch and, at the appropriate positions of the switch, are fed directly to the two-stage preamplifier-equalizer circuit built around V3-A and V3-B, a 12AX7. In the PHONO position a bass-boost circuit is connected between V3-A and V3-B; the turnover frequency and amount of boost are determined by the position of TURNOVER control SW-2. There are four turnover positions: Old LP, RIAA, Old AES, and Early 78 (800-cps turnover). This control is at the extreme left on the front panel, Fig. 3. In the MIC position

Fig. 1. Circuit of the Heathkit WA-P2 preamplifier-equalizer-control unit.



COURTESY HEATH COMPANY

of the selector switch, the bass boost circuit is shorted to ground.

Treble rolloff is accomplished by shunt capacitors in the plate circuit of V3-B. The amount of rolloff is determined by the ROLLOFF control, SW-2, which also has four positions: 0, 8, RIAA-12, and 16. The figures indicate, of course, the number of db that response is down at 10 Kc relative to 1 Kc. This control is second from the left in Fig. 3. It is removed from the circuit in the MIC position of the selector switch. Level controls for both the PHONO and MIC inputs are located in V3-B's plate circuit, and these are fed to the same wafer on the selector switch as are the high-level inputs.

The selected input is then fed to V1-A, half of a 12AU7, which is connected as a cathode follower. Its output at low impedance goes to the TAPE output jack on the rear panel, and also to V2-A. This stage, half of another 12AX7, functions as an amplifier and driver for the tone control circuits. Bass and treble controls are individual continuously variable losser circuits. They are the two small knobs in the center of the front panel, Fig. 3. The treble control has a switch, activated in its maximum counterclockwise position, that can be used to turn AC power on and off. V2-B, another straight amplifier stage, follows the tone controls and is followed by the main volume control. Output of the volume control is fed to V1-B, the remaining half-12AU7, which is connected also as a cathode follower. This feeds, at low impedance, the main output jack on the rear panel.

Location of the volume control is at the far right on the front panel; the selector switch is just to its left.

Fig. 3. Front panel and main controls. Plug on cable connects to a power source.

Fig. 2. Back panel of the preamp, showing inputs, outputs, and level controls.

The WA-P2 does not have its own power supply. It is equipped with a long power cable and an octal plug; all Heath amplifiers have sockets that will accept the plug, and these sockets are wired to furnish the proper operating voltages for the preamp. Many other amplifiers have similar power outlet sockets, and most of those that do not have the socket have enough reserve to supply the relatively small requirements of the WA-P2. A socket can usually be installed somewhere on the amplifier chassis, and connections made to it in accordance with directions in the WA-P2 instruction book. The power requirements are 6.3 volts AC at 1 amp, and 300 volts DC at 10 ma. The high-voltage supply is very well decoupled in the preamp, so there is virtually no danger of motorboating because of feedback in the filtering system. There must be no external connection between the filament supply and ground; if the filament winding in the amplifier's power transformer has a grounded center-tap, that connection must be removed. This is because there is a hum-bucking potentiometer across the filament circuit in the preamp (accessible through the bottom cover).

Construction Notes

Clever design has reduced the complexity of the WA-P2 for inexperienced builders by providing for a number of subassemblies that are wired separately. These are shown in Figs. 4 and 5. Front or top views of the chassis, front panel, left end bracket, and rear panel are given in Fig. 4; the opposite sides of these subassemblies are shown in Fig. 5. Chassis wiring has not yet been done in these views.

After the individual units are completed they are assembled as shown in Fig. 6, the top view, and in Fig. 7, as seen from the bottom. Connections are then made between them. The chassis is attached at its ends to the end brackets, by means of rubber shock mounts which tend to reduce the preamp's susceptibility to shocks and microphonic feedback. After interwiring and testing, the cabinet shell and bottom plate are installed. Final appearance is that shown in Figs. 2 and 3. The bottom plate has rubber feet so that the preamp can be used out in the open, and set on furniture without scratching. It also has an access hole for adjustment of the hum control (framed by two capacitors at the right end of the chassis, Fig. 7).

Several of these units have been built by various members of our staff during the past year. All have worked perfectly at completion, and there have been no failures in use — a tribute to the basic design, the thoroughness and accuracy of the assembly instructions, and the quality of the parts furnished. It should be emphasized, however, that such results can be expected only if the instructions are followed to the letter and without error. There are 197 numbered steps, and a mistake in any one of them could negate the most minute care taken everywhere else. It took us 14 hours from start to finish on the unit that is the subject of this article; we might have done it in less time, but then we might have made an error too. If you don't try to work on a kit such as this when you're very tired, and don't rush the job, the chances are excellent that it will work perfectly the first time you turn it on.

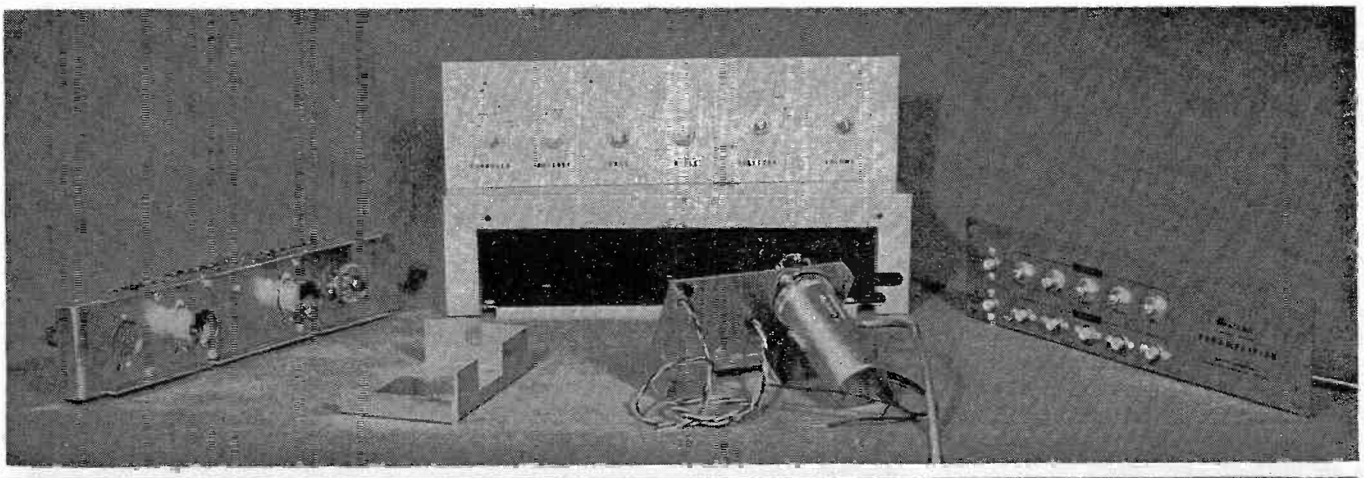
Other than those above, we have no specific comments on the kit's construction. The instruction book has, seemingly, covered everything.

AUDIOCRAFT Test Results

Four full pages of the instruction book are given to the most exhaustively complete specifications we recall seeing for any similar unit. We don't have room to publish this section in its entirety, but we shall give a few of the more conventional specifications:

Gain. (Input level and volume controls set at maximum). On high-level inputs, .09 volt input produces 1 volt output; on low-level inputs, 2.5 mv input produce 1 volt output.

Frequency response. (MIC input, tone



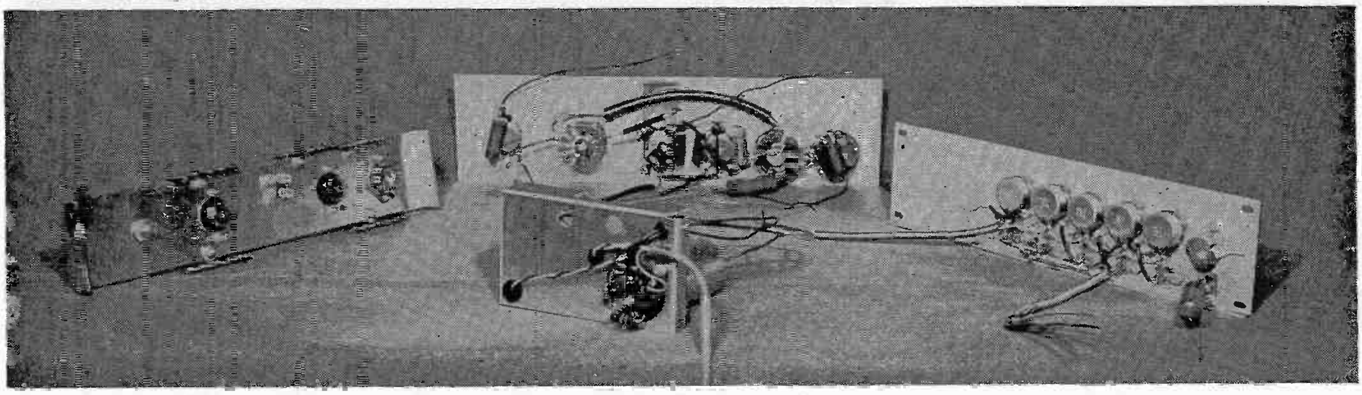


Fig 4, above, and Fig. 5, below: views of the individual subassemblies. Wiring of chassis (at the left) is not yet done.

controls set for flat response at 100, 1,000, and 10,000 cps). Within 1 db from 25 cps to 30 Kc; within 1.5 db from 15 cps to 35 Kc.

IM distortion. (60 and 7,000 cps, 4:1, tone controls flat, volume control at maximum, output level controlled by input level control). At 1 volt output, 0.5% on MIC input, 0.3% on high-level inputs: at 2 volts output, 0.55% on MIC input and 0.59% on high-level inputs.

Hum and noise. (Tone controls set for flat response, TURNOVER control on LP, ROLLOFF control on 0, power cord polarized and hum balance control adjusted for minimum hum, input level controls adjusted for 2.5 volts output at input voltages shown.) For 0.5 volt at TUNER input, 72 db below 2.5 volts; for 6 mv at PHONO input, 62 db below 2.5 volts; for 15 mv at MIC input, 70 db below 2.5 volts.

Tone controls. Bass control, continuously variable from 18 db boost to 12 db cut at 50 cps; treble control, continuously variable from 15 db boost to 20 db cut at 15 Kc.

Dimensions. (Over-all) 12 9/16 in. long by 3⅝ high by 5⅞ deep.

These are stated to be measurements made on a typical kit, and that minor deviations were to be expected because of parts tolerances and the wide variations in dress of parts and wiring. There were indeed some deviations in our kit, evident from our test results, but in most cases they were in a favorable direction. Fig. 8, for instance, shows the response from a high-level input to the main output jack, with a 4-ft. shielded cable connected to the output jack and a load of 500 K. The center curve is response with both tone controls set flat; the upper curve is relative response when both tone controls were set to maximum boost; the lower curve is that for maximum-cut setting of both controls. At 50 cps the range was approximately +17.5 to —15 db. At 15 Kc the range was about +17 to —17.7 db. Note, however, that the middle-range response is boosted or cut slightly as well as the frequency extremes; this would reduce the effective range of the controls by a small amount. These are controls of the "hinge" type, in which the slope — not the turnover frequency — is varied as the control is turned. There is some disagreement as to which type is better, but there can be no argument that the curves in Fig. 8 are quite adequate in range and symmetry.

Curves for the preamplifier-equalizer section are shown in Fig. 9. It can be seen that equalization continues down to about 30 cps, and that the maximum boost is 22 db. Both are indicative of high-quality design. The RIAA playback curve indicates somewhat more bass boost, and less treble cut, than that

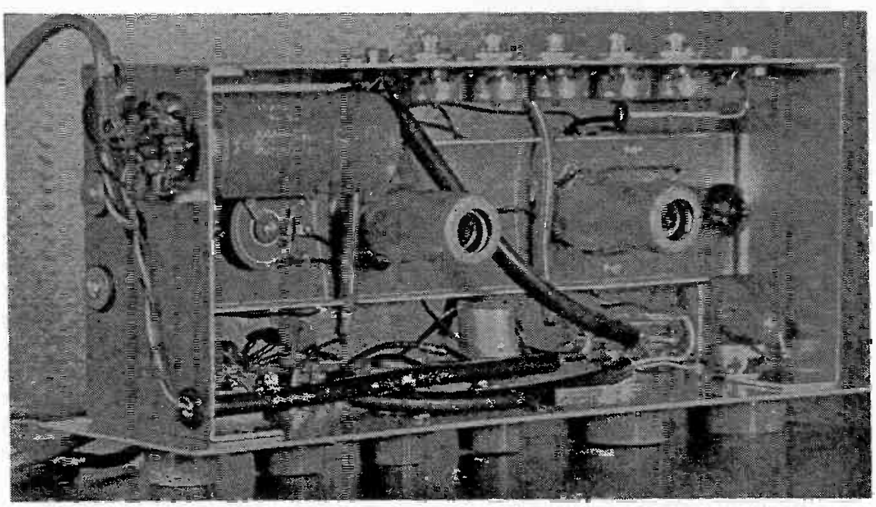


Fig. 6. Completed preamp with its outer shell removed. Note shock-mounted chassis.

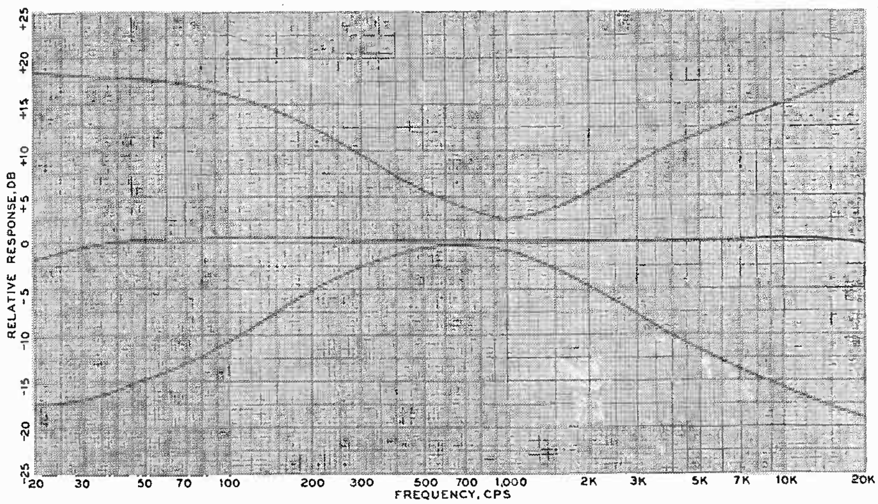


Fig. 8, right: operating ranges of tone controls, and response in flat settings.

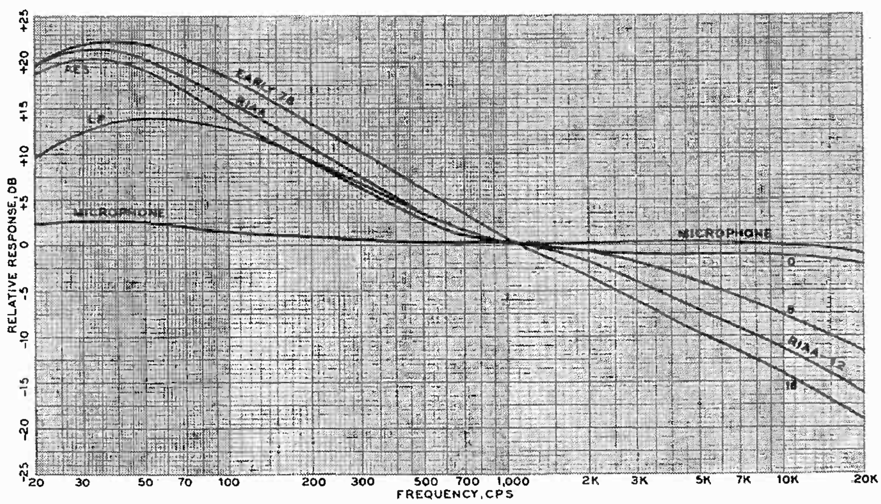


Fig. 9, below at right: response from low-level input jacks to main output.

required by the recording curve, but the error is small enough to be corrected easily by the tone controls if that is felt to be necessary.

Just about every popular type of pickup cartridge has been used with this preamplifier, and with none has hum become objectionable. With most it was completely inaudible. Gain is phenomenally high, too; at one time we used an Electro-Sonic pickup without its transformer, and had no trouble getting as much noise-free volume as we could use.

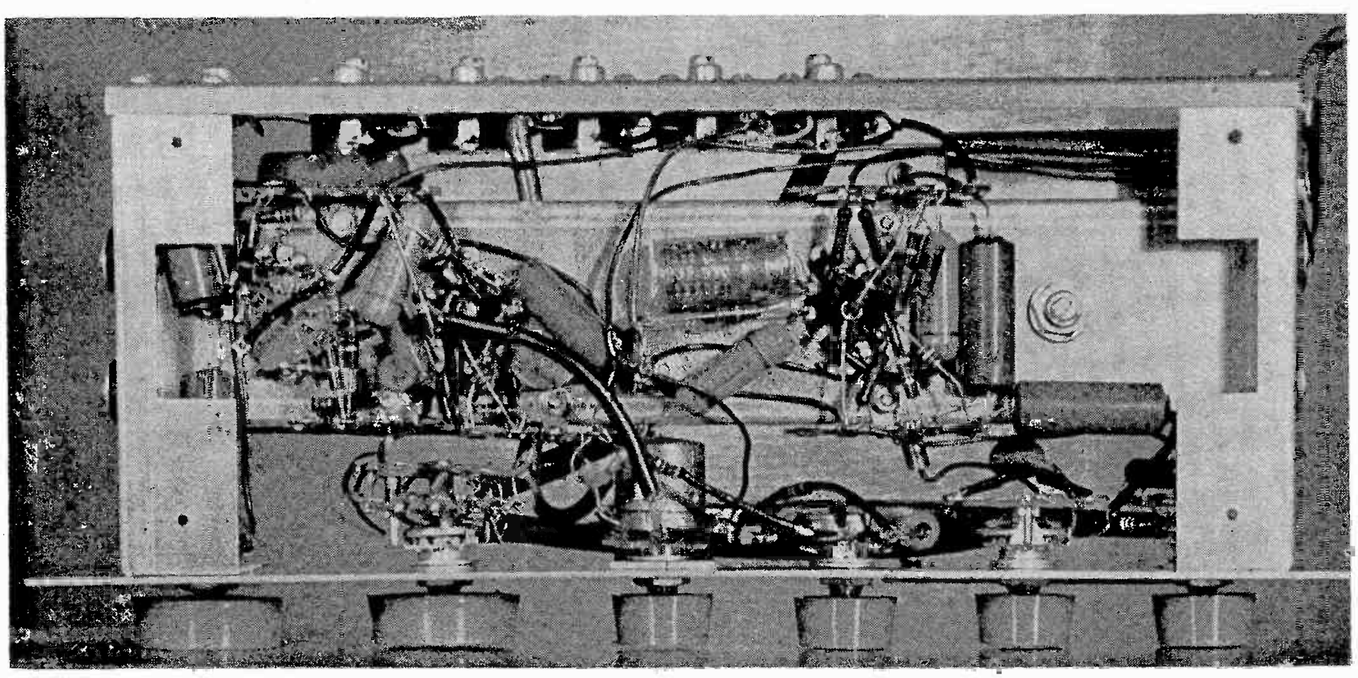
In the matter of distortion, we found some very good and some not-so-good aspects. This preamplifier is capable of as fine performance as anyone could desire, and about as fine as you can get. But you have to use it properly; that is to say, you have to keep the input level controls turned down as far as possible, and the main volume control as far up as possible.

The reason is, simply, that the volume control is far toward the output end of the circuit. Both the preamplifier and control sections have relatively high gain. If you feed a normal signal in, and keep the input level control up high, the signal levels around the tone control stages become very high indeed. The level at the output jack can be reduced by the volume control, to be sure, but reducing the level at that point cannot reduce distortion introduced by the overloaded preceding stages. Figs. 10 and 11 reveal clearly the results of this process.

Performance on high-level inputs is shown in Fig. 10. Distortion (IM, 60 and 7,000 cps mixed 4:1) is plotted for output levels up to 2 volts, with a curve for each of four volume-control positions. With the volume control ¼ on, distortion at 1 volt output is just over 6%; it is well over 30% for 1.5 volts output. This is reduced substantially with the volume control at ½ rotation — to 1.6% and 4% for

Continued on page 36

Fig. 7. Bottom view with cover plate and shell off. Chassis wiring is shown here, and also connections between subassemblies.



BASIC ELECTRONICS

by Roy F. Allison

VIII: Parallel and Series Reactances

THE preceding two chapters have been concerned with the behavior of single inductors and capacitors in circuits with resistance. In this chapter we shall assume the same circuit conditions, except for replacement of a single inductor or capacitor by combinations of two or more. In other words, the resultants of inductors and capacitors in series and in parallel connections will be examined.

Inductors in Series

Let us suppose that a number of inductors are connected in series, as shown in Fig. 1. Three are shown, although the

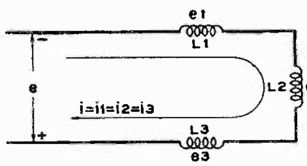


Fig. 1. Series inductors add directly.

number could be any; the rest of the circuit is not shown, since it is unimportant to this discussion. The voltage e is the total voltage that appears across this section of the circuit. A lower-case notation is used to indicate that this quantity may be changing from moment to moment — it is the instantaneous value we are dealing with.

Now, the most significant apparent characteristic of an inductor is that it resists rapid changes in current. It will be recalled from Chapter VI that, when a sudden change of voltage was applied to an inductor and a resistor in series, current in the circuit built up gradually, and the entire voltage was dropped across the inductor at the first instant. It took time to build up the magnetic field about the choke. If 1 choke resists changes in current to a certain degree, then logic tells us that 2 chokes of the same size, connected in series so that the same current must force its way through both, will resist the change in current just twice as stubbornly. This resistance to *changes* in current, called inductive reactance, will be trebled if 3 chokes of the same size are connected in series, and so on. It follows that the total inductance of inductors in series is the sum of the individual inductance values.

This rule can be verified easily by simple mathematics. It will be recalled that the relation among inductance, voltage across the inductance, and the rate of change of current in the inductance is given by the formula

$$E = \frac{Ldi}{dt}.$$

Let us call the total instantaneous voltage in Fig. 1 e, the total or resultant inductance in the circuit L, and the instantaneous circuit current i. The voltage across L1 is identified as e1; that across L2, e2; and that across L3, e3. Then:

$$e1 = \frac{L1di}{dt},\ e2 = \frac{L2di}{dt},\quad e3 = \frac{L3di}{dt},$$

from the fundamental equation. In the same way,

$$e = \frac{Ldi}{dt}.$$

But, also, $e = e1 + e2 + e3$. Thus

$$\frac{Ldi}{dt} = \frac{L1di}{dt} + \frac{L2di}{dt} + \frac{L3di}{dt},$$

$$L\left(\frac{di}{dt}\right) = L1\left(\frac{di}{dt}\right) + L2\left(\frac{di}{dt}\right) + L3\left(\frac{di}{dt}\right),$$

$$L\left(\frac{di}{dt}\right) = (L1 + L2 + L3)\left(\frac{di}{dt}\right)$$

Dividing both sides by (di/dt), we obtain

$$L = L1 + L2 + L3;$$

that is, total inductance is equal to the sum of the individual inductances. This is the same as for resistances in series. Moreover, from the rearranged basic expression

$$e = L\left(\frac{di}{dt}\right),$$

and from consideration of the fact that the (di/dt) term is identical for all inductors in the series circuit, it is apparent that the instantaneous voltage drops are directly proportional to the individual inductance values. To illustrate: assume for convenience a total instantaneous voltage of 10 volts, and inductors of 2, 3, and 5 h for L1, L2, and L3, respectively. The total circuit inductance is 10 h. According to the formula, (di/dt) must equal 10 volts divided by 10 h, or 1 ampere per second, and

$$e1 = 2 \times 1 = 2 \text{ volts};$$
$$e2 = 3 \times 1 = 3 \text{ volts};$$
$$e3 = 5 \times 1 = 5 \text{ volts}.$$

Voltage division among inductors in series, then, is directly proportional to the individual values of inductance.

Inductors in Parallel

Let us continue working with three inductors, but connect them in parallel rather than in series. The new circuit is given in Fig. 2. This is obviously going to change the situation because now the same voltage will be across all the inductors, and the currents will be different in each.

If only L1 were connected in the circuit it would resist changes in current to a degree determined by its own inductive reactance. Now, adding L2 in parallel would furnish an alternative path for circuit current; assuming that the voltage at the source was not affected by the addition of L2, the rate of current change in L1 would not be altered. The *total* rate of current change in the circuit, then, would be increased by the addition of L2. Accordingly the total reactance — and the total or resultant inductance — must have decreased. If L1 and L2 were equal in value, the total rate of current change in the circuit would be doubled over that with

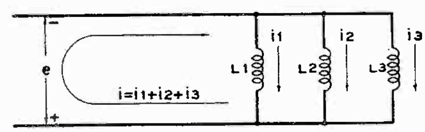


Fig. 2. Rate of current change through each inductor is determined individually.

L1 alone, so that the total inductance would have been halved. With three equal inductances in parallel, the total inductance would be equal to ⅓ that of a single unit.

All this is very well, but what is the relation for parallel inductors of unequal value? Again, some simple mathematics will tell us. We shall work with three inductances, but it should be remembered that this derivation is valid for any number. The instantaneous current in L1, Fig. 2, will be called i1; that in L2, i2; and that in L3, i3. Total instantaneous current is simply i, and total inductance, L. The instantaneous voltage e is common for all branches and for the total circuit. Accordingly,

$$e = \frac{Ldi}{dt},$$

and, by transposition,

$$\frac{1}{L} = \frac{di}{edt}.$$

$$\text{But } \frac{di}{dt} = \frac{di1}{dt} + \frac{di2}{dt} + \frac{di3}{dt}.$$

Substituting this term for di/dt,

$$\frac{1}{L} = \frac{di1 + di2 + di3}{edt}, \text{ or}$$

$$\frac{1}{L} = \frac{di1}{edt} + \frac{di2}{edt} + \frac{di3}{edt}.$$

Working with the individual branches now,

$$e = \frac{L1di1}{dt};\ e = \frac{L2di2}{dt};\ e = \frac{L3di3}{dt}.$$

Transposing terms,

$$\frac{1}{L1} = \frac{di1}{edt};\ \frac{1}{L2} = \frac{di2}{edt};\ \frac{1}{L3} = \frac{di3}{edt}.$$

The second term is equal in each case to one of those in the final formula for the total circuit above. Substituting equivalencies,

$$\frac{1}{L} = \frac{1}{L1} + \frac{1}{L2} + \frac{1}{L3}, \text{ or}$$

$$L = \frac{1}{\frac{1}{L1} + \frac{1}{L2} + \frac{1}{L3}}.$$

This is identical in form with the equation for the resultant of resistances in parallel. In the same way, when only two inductors are involved, the equation reduces to a simpler form:

$$L = \frac{L1\ L2}{L1 + L2}.$$

If more than two inductors must be considered in the calculation the simpler formula can, as with resistances in parallel, be applied to any two. Then, treating the resultant value as one inductance, the resultant of this and the remaining inductance can be found.

As an illustration, assume the same values as before for L1, L2, and L3: 2, 3, and 5 h, respectively. The resultant of L1 and L2 is

$$\frac{2 \times 3}{2 + 3} = \frac{6}{5} = 1.2 \text{ h}.$$

The resultant of 1.2 h and L3 is

$$\frac{1.2 \times 5}{1.2 + 5} = \frac{6}{6.2} = \text{approx. } 0.97 \text{ h}.$$

This can be checked by assuming an arbitrary voltage for the circuit. Let us take 10 volts for purposes of calculation. Then,

$$\frac{di1}{dt} = \frac{e}{L1} = \frac{10}{2} = 5 \text{ amps/sec.}$$

$$\frac{di2}{dt} = \frac{e}{L2} = \frac{10}{3} = 3\tfrac{1}{3} \text{ amps/sec.}$$

$$\frac{di3}{dt} = \frac{e}{L3} = \frac{10}{5} = 2 \text{ amps/sec.}$$

This gives us the total rate of current change in the circuit for an applied voltage of 10 volts:

$$\frac{di}{dt} = \frac{di1}{dt} + \frac{di2}{dt} + \frac{di3}{dt}$$

$$= 5 + 3\tfrac{1}{3} + 2 = 10\tfrac{1}{3} \text{ amps/sec.}$$

By transposition of the basic formula,

$$L = \frac{edt}{di} = \frac{10}{10\frac{1}{3}} = \text{approx. } 0.97 \text{ h.}$$

It should be noted that the smallest inductor permitted the most rapid current change and, accordingly, had most effect on the total inductance. The current divided in inverse proportion

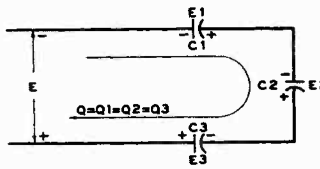


Fig. 3. Capacitors have different rules.

to the inductance values, as it does for resistances in parallel. If L2 and L3 had been 50 times larger in inductance than L1, they would have had but little effect on the total inductance. Thus, the total inductance of inductors in parallel is always less than the smallest, and is determined to a greater extent by the smallest than by any of the others. This is exactly opposite to the series connection, in which the largest inductor has most effect.

The chart given in Fig. 8, Chapter IV, for the resultant of resistances in parallel, can be used to find the resultant of inductances in parallel also. Operation is the same as for resistances.

Capacitors in Series

Alternative forms of the equation relating capacitance, charge, and voltage, it will be recalled, are

$$E = \frac{Q}{C},\quad C = \frac{Q}{E}, \text{ or } Q = CE.$$

Capacitance is a measure of ability to act as a reservoir of charge. The greater the capacitance of an object, the greater quantity of charge it will assume with a given applied voltage.

With these facts in mind let us examine the circuit in Fig. 3. Three capacitors are shown connected in series; it could be more or less, but it is convenient to work with three. The common element in this circuit is the charge, which must be the same on all three capacitors. This is a vital point: the charge for any capacitor cannot flow *through* the other capacitors. Charge electrons on the negative plate of C2, for example, can come only from the positive plate of C1, and the electrons that leave the positive plate of C2 can go only to the negative plate of C3. So on around the circuit; because each capacitor (for practical purposes) blocks continuous current in the circuit, the quantity of charge has to be equal on each.

Now we must proceed with care. The voltage at the source will be divided among the three capacitors because this is a series circuit. According to the first formula above, the voltage across any capacitor is directly proportional to the quantity of charge and inversely proportional to the capacitance. Here the charges are all equal; therefore, the source voltage must be distributed in inverse proportion to the values of capacitance! The largest capacitor must have the smallest voltage across it, and the smallest capacitor the largest voltage. This is exactly opposite to what happens with resistances and inductances. It can't be right, can it? Next we'll be finding that the total capacitance isn't the sum of the individual capacitance values, but something smaller than any of them.

Well, that's just what it is. Let's look at it this way: suppose the capacitors are all equal in value and identical in physical dimensions. We shorten the connecting wires between them to insignificant lengths, so that they are effectively stacked plate-to-plate, one above another. We now have the equivalent of a single capacitor whose plates are separated by three times the distance of one of the original capacitors. Since the capacitance of a parallel-plate capacitor is inversely proportional to the spacing between the plates, the total capacitance of our stack of three capacitors is ⅓ that of one of them.

It may make even more sense when seen from another viewpoint. If the total source voltage were impressed across 1 of the 3 capacitors a certain charge would flow — say, 6 coulombs. By putting 3 capacitors of the same value in series, the voltage across each is reduced to ⅓ that of the source. Then, because charge is directly proportional to the voltage across a capacitor, there would be only ⅓ the charge flow in the circuit, or 2 coulombs. With the same voltage at the source and only ⅓ the charge flow, obviously the total capacitance has been reduced to ⅓ that of a single capacitor.

The exact relationship for odd values can be found as follows: referring to Fig. 3, the voltage across C1 is E1; that across C2, E2; and that across C3, E3. The charge flow, Q, is the same for each circuit element and for the circuit as a whole. Source voltage is E. The total capacitance, then, is

$$C = \frac{Q}{E};\ \text{or}\ \frac{1}{C} = \frac{E}{Q}.$$

But $E = E1 + E2 + E3$. Substituting this for E,

$$\frac{1}{C} = \frac{E1 + E2 + E3}{Q}, \text{ or}$$

$$\frac{1}{C} = \frac{E1}{Q} + \frac{E2}{Q} + \frac{E3}{Q}$$

For the individual capacitors,

$$C1 = \frac{Q}{E1};\ C2 = \frac{Q}{E2};\ C3 = \frac{Q}{E3};\ \text{or}$$

$$\frac{1}{C1} = \frac{E1}{Q};\ \frac{1}{C2} = \frac{E2}{Q};\ \frac{1}{C3} = \frac{E3}{Q}.$$

The second terms in each case are identical to terms in the right-hand section of the modified formula for total capacitance above. Substituting, we obtain

$$\frac{1}{C} = \frac{1}{C1} + \frac{1}{C2} + \frac{1}{C3},\ \text{or}$$

$$C = \frac{1}{\frac{1}{C1} + \frac{1}{C2} + \frac{1}{C3}}.$$

Thus, the formula for the resultant of capacitors in *series* is equivalent to those for resistors and inductors in *parallel*. By the same token, it can be simplified through application of its reduced form for two capacitors in parallel:

$$C = \frac{C1\,C2}{C1 + C2}.$$

Further, the chart for finding resultants of resistors in parallel can be used for capacitors in series.

Capacitors in Parallel

Suppose we rearrange the three capacitors to parallel connection, as shown in Fig. 4. Now the voltage across each is that of the source, and the charge on each, flowing in an individual path, is determined by its capacitance value and the voltage. Let us assume specific values: say that C1 is 2 μfd, C2 is 4 μfd, and C3 is 10 μfd. Call E 10 volts. The charge on C1 will be $2 \times 10 \times 10^{-6}$ or 20×10^{-6} coulombs; that on C2 will be $4 \times 10 \times 10^{-6}$, or 40×10^{-6} coulombs; and the charge on C3 will be $10 \times 10 \times 10^{-6}$, or 100×10^{-6} coulombs.

Now, the total charge flow through the source is the sum of the individual charges, $(20 + 40 + 100) \times 10^{-6}$, or 160×10^{-6} coulombs. To the source, then, the three capacitors in parallel appear as

$$C = \frac{Q}{E} = \frac{160 \times 10^{-6}}{10} = 16\ \mu\text{fd}.$$

This is the sum of the three individual capacitance values. It appears that the resultant value of capacitances in parallel is, simply, the sum of the individual values. Is this readily explicable? Let us go back once again to the physical concept of a capacitor. Assume that the three capacitors we have just been working with have identical plate spacing and dielectric, which are set up so that each square foot of plate area results in 1 μfd of capacitance. C1 would then have plates 2 sq. ft. in area, C2 would have plates 4 sq. ft. in area, and C3 would have plates 10 sq. ft. in area. Connecting the capacitors in parallel joins the three top plates electrically and the three bottom plates electrically; it has the same electrical effect as creating a larger capacitor whose plate area is the sum of the individual plate areas. This would be 16 sq. ft. and, since capacitance is directly proportional to plate area, the total capacitance would be 16 μfd — the sum of the individual capacitances.

The mathematical proof is straightforward and simple. As we have done before, let us call C the total, or resultant, capacitance. Q1 is the charge on C1, Fig. 4; Q2 is the charge on C2; and Q3 is the charge on C3. E is the source voltage, common to the individual units and the resultant. Then, for the resultant,

$$C = \frac{Q}{E}.$$

Q is the sum of $Q1 + Q2 + Q3$, because the individual charges all flow through the source. Accordingly,

$$C = \frac{Q1 + Q2 + Q3}{E},\ \text{or}$$

$$C = \frac{Q1}{E} + \frac{Q2}{E} + \frac{Q3}{E}.$$

For the individual capacitors

$$C1 = \frac{Q1}{E};\ C2 = \frac{Q2}{E};\ C3 = \frac{Q3}{E}.$$

The second terms all appear in the resultant equation just above. Substituting their equivalents in that equation,

$$C = C1 + C2 + C3.$$

This is in the same form as the equations for resistors and inductors in *series*. Further, it can be seen that the total capacitance of a parallel group of capacitors is always *larger* than that of any individual capacitor in the group, and that the *largest* capacitor in the group has most effect on the total capacitance. That is exactly opposite to the rules for *parallel* resistors and inductors, and precisely the same as those for *series* resistors and inductors. Readers confused at this point have one consolation: the rules for capacitors are consistent in their inconsistency.

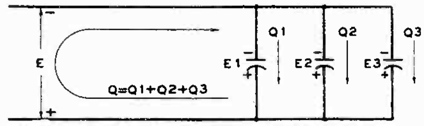


Fig. 4. Resultant of capacitors in a parallel connection is the sum of all.

The table below gives condensed information on the characteristics of all three types of circuit element, in series and in parallel circuits.

	RESISTORS		INDUCTORS		CAPACITORS	
	SERIES	PARALLEL	SERIES	PARALLEL	SERIES	PARALLEL
Common Property	Current	Voltage	Rate of current change	Voltage	Charge flow	Voltage
Individual Property	Voltage	Current	Voltage	Rate of current change	Voltage	Charge flow
Resultant Value of Two or More	Sum of individual values	Less than smallest value	Sum of individual values	Less than smallest value	Less than smallest value	Sum of individual values
Voltage Division	Directly prop. to individual values	Same for all	Directly prop. to individual values	Same for all	Inversely prop. to individual values	Same for all
Current or Charge Division	Same for all	Inversely prop. to individual values	Same for all	Inversely prop. to individual values	Same for all	Directly prop. to individual values
Important Distinguishing Features	Permits abrupt voltage or current changes. Consumes energy as heat. Does not store energy.		Permits abrupt voltage changes. Does not permit abrupt current changes. Does not consume energy. Stores energy in magnetic field.		Permits abrupt current changes. Does not permit abrupt voltage changes. Does not consume energy. Stores energy in electrostatic field.	

TEST INSTRUMENTS

Continued from page 22

Since the meter responds only to DC, this portion of the output signal is rectified by a half-bridge using crystal diodes. Nonlinearity of the diodes at low signal levels is compensated by a third diode across the meter. When the audio signal generator is operated with the proper termination, the meter and attenuator will indicate the output level at the binding posts. Metered output is especially valuable for audio tests in which constant output of the generator is essential at various frequencies. The output control can be varied if necessary to give the same output level at all frequencies within the range of the instrument.

This particular audio signal generator was selected as an example of modern generator design because of the features it incorporates. Step-type tuning provides for accurate setting to a given frequency, especially valuable in frequency-response checks. This circuit provides exceptionally low distortion at the output (less than 0.1% from 20 to 20,000 cps), and metered output combined with both continuously variable and step-type attenuators with selection of internal or external load. Such an instrument is tailor-made for high fidelity audio work, and an understanding of its internal operation will be helpful in employing it to maximum advantage. Understanding how a piece of test equipment works is really the first step in determining how it should be used.

In the next installment we will pursue the subject of audio signal generators further, and discuss some of the mechanics of using such an instrument. A procedure for using this generator and a VTVM for making frequency-response runs on high fidelity equipment will be described.

MUSIC FROM ELECTRONS

Continued from page 19

(the triode section of V8), similar in design to the variable-pitch oscillator, is constructed so that changes in capacitance between the left hand and the volume antenna change the frequency of oscillation. A signal from this oscillator is fed into a narrow-band RF amplifier (the pentode section of V8) so that, as the frequency of oscillation changes, the amplitude of the output of the RF amplifier also changes. This output is rectified, and the resultant potential is applied to the control grid of the control amplifier V6a. When this bias is made negative, the gain of V6a decreases; when the bias is made positive, the gain of V6a increases. By varying the capacitance of the volume antenna with the left hand, the performer is able to make the tone loud or soft, or to silence it altogether. From the control amplifier, the signal is fed into a cathode follower, and then to an external amplifier and loudspeaker.

The power supply is conventional in design. A glow voltage regulator tube is used to supply constant plate voltage to the oscillators, so that variations in line voltage do not cause variations in pitch or volume while the performer is playing.

Theoretically, any type of music can be played on the theremin. Both pitch and volume-control circuits respond instantly to changes in hand capacitance. Melodies can be played as fast as the hands can move. In practice, however, the theremin has proved itself best adapted to slow melodies which give the hands time to locate themselves accu-

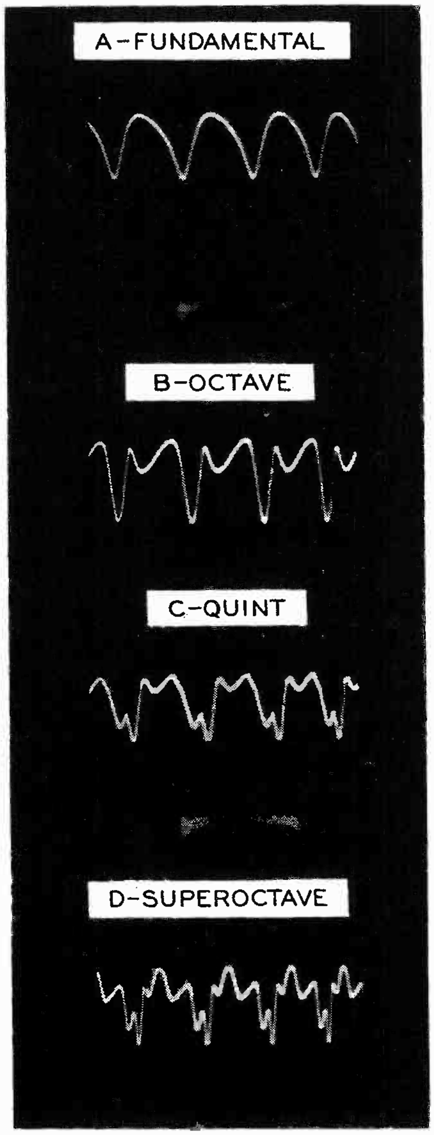


Fig. 6. Harmonics added to the original.

rately on each note. Slow melodies, in addition, lend themselves to the use of vibrato. The thereminist's greatest asset is his ability to impart a beautiful vibrato to the tone merely by moving his right hand back and forth through a small distance.

In this age of network broadcasting, tape and disc records, and high fidelity sound, music has enriched the lives of millions of people. The theremin was the first electronic instrument that generated, rather than reproduced, music. As such it has pioneered the field of electronic musical instruments. Because of his knowledge of electronics and appreciation of music, the high fidelity enthusiast will find the study of electronic musical instruments an interesting and worthwhile pursuit.

TAPE NEWS

Continued from page 13

the preceding program, wait through a couple of peaks to be sure the control engineer isn't already taking care of things.

Also, while recording, try to refrain from pulling up the volume of low-level passages despite the fact that nothing is registering on the record indicator. Contrary to what is apparently a very popular belief, "normal" recording volume does not mean that all of a program must necessarily be recorded at that level. Set for normal volume on the high-level passages, and let the low-volume ones take care of themselves.

The final step, once the recording is completed, is to edit the unwanted sections out of the program. In the category of "unwanted" things, I would list commercials, tedious introductory commentaries, and any loud coughs that may find their way into the program.

As an example of what might be done with a splicer and a roll of editing tape, let's take a look at a typical broadcast of a symphony orchestra over a typical non-good-music station. (I'm choosing typical examples because they are typical of the typical situation encountered by the typical recordist).

Our typical program is a live broadcast of the Hinckleyville Symphony Orchestra, coming to us from Hinckleyville's FM mediocre-music station. The program is preceded by Jed Barnstörm's Old Original Amateur Hour, which may be classed as musical in a broad sense, so it allows us to set the record indicator to read NORMAL on program peaks.

Next is a commercial for Mama Mia's Spaghetti, which is speech so it doesn't push the recording indicator past the half-way mark. But we are not fooled; we let the volume setting stay where it is.

Then comes an unidentified announcer who verifies the program listing in the newspaper — we are, indeed, about to hear the Hinckleyville Symphony Orchestra. At this point, just to be cautious, we start the tape recorder running, and are glad we did because the orchestra plunges into an overture without further ado. This is later identified as the *Poet and Peasant* Overture, and is followed by a commercial. Next is a noble attempt at Ravel's Second *Daphnis and Chloe* Suite, at which point there is another commercial and our tape runs out. We are not surprised, because we have noted from an old record catalogue that *La Valse* occupies three 12-inch sides, and figuring 4½ minutes per side we estimated 17½ minutes for this one, at the very most. The *Poet and Peasant* usually occupied

Continued on next page

Stylus-Use Timer

It is important to keep track of the number of hours a stylus has been in use, so that it can be replaced before it has become worn. By the time distortion is audible the stylus may have done some damage to record grooves. It is also desirable to know the length of time a turntable has been run; many require lubrication every thousand hours or so.

I have discovered a very easy way to keep track of the time a stylus or turntable has been in use. I connected a Running-Time Meter in parallel with my turntable (on the turntable side of the AC switch). This is easily done on the Rondine Jr., and I imagine the switch is more or less accessible on all turntables and most changers. The meter records the number of hours the turntable has been run, and it also gives a good approximation of the number of hours the stylus has been in use. By noting the meter reading when a new stylus is installed, or when the turntable is oiled, an accurate record can be kept with a minimum of effort.

My meter is made by Industrial Timer Corporation. It is Model C2 and cost $15.50. It registers up to 9,999.9 hours and then starts over. A similar model which can be reset to zero costs $35. The meter is compact and easy to mount; it is about three inches in diameter and not bad-looking, as meters go.

One word of caution: the meter makes some noise while running, like an electric clock, and puts out a powerful electric field. It should be mounted far away from electronic components to avoid hum pickup.

Colston Nauman
Chicago, Ill.

Amplifier Auto-Switch

Mr. Thayer, in his Audio Aid suggestion (AUDIOCRAFT, February), outlined a very useful method for turning his amplifier on and off by the switch on his Garrard changer. However, its application is limited to using the amplifier only with the changer. By use of a relay, the amplifier can be controlled with either tuner or changer.

The 110-volt AC coil of the relay can be paralleled across the changer motor. The center connections on the DPDT relay switch (an SPDT can also be used, but with different wiring) go to the amplifier. The side of the switch closed when the coil is energized connects directly to 110 volts AC; the other side connects to the switched outlet on the tuner.

The amplifier will now go on and off with either the tuner or changer. Turning on the tuner while stacking the records allows the amplifier to warm up.

Clinton D. Nelson
Park Forest, Ill.

Speaker Isolation

Here is an idea that may be of interest to owners of speaker systems having a six-inch cone mid-range speaker. It is desirable to isolate the mid-range speaker from the woofer in order to eliminate any possible intermodulation effects. While a one-foot-square box is quite effective, its use is not always desirable because it occupies a cubic foot of the cabinet, and makes it difficult to get at the speaker.

An effective alternative is a layer of two-inch Kimsul over the six-inch unit, and the whole thing covered with a man's old felt hat. The shape of the hat fits the contour of the unit, and the edges of the hat may be tacked down flush with the panel to form a tight seal. Apart from a solution of the problem of what to do with your old hat, this will also give better sound from your speaker system.

Albert Sadler
San Diego, Calif.

AUDIO AIDS WANTED

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

TAPE NEWS

Continued from preceding page

two sides, so 27 minutes was estimated for the first half of the program.

Since our recorder has fairly slow rewind, we finish running the tape onto the takeup reel (to save time) and swap the empty feed reel over to the takeup side, putting a fresh reel on (if we have a full-track recorder) or putting the full takeup reel on for the return trip. Since the intermission took us slightly past the half-hour mark on timing, we may be sure our reel won't run out during the second half of the program (Ravel's *Valses Nobles et Sentimentales*). Had the hour-long program been less conveniently divided, we would have been obliged to use two reels of tape, changing reels during the middle commercial, or we would have had to use a reel of double-play tape. For those of us with unlimited financial resources, a 10½-inch reel capacity or a second recorder to switch over to would have been the answer.

After the program is over, we fade out the volume during a burst of applause and prepare to edit the recording.

First goes the tail end of the Mama Mia Spaghetti commercial, along with the announcement and commercial following the overture. Likewise the other bits of addenda that go to make up the usual radio broadcast.

Then to the programming. First of all, the long bits of tape edited from the broadcast are spliced together and run through the recorder, to erase them. Then they are used to add spaces of silence between the recorded items on the tape. The amount of time left between each selection and the preceding one will determine to a large extent the pacing of the program; shorter pauses give the whole program a sort of frenzied atmosphere, while a longer pause induces relaxation. The average time between two selections should run to around 8 seconds, with 5 or 6 seconds between movements of a single work.

When editing orchestral tapes, be sure not to lop the end off the reverberation following the music, and *never* cut right into the middle of applause. Wait until the applause has died out and then end the recorded section there, following it by the necessary amount of blank tape to bring the pause between musical numbers up to its 8-second duration. Applause should be counted as a pause in this connection unless it runs on incessantly, in which case it should have been faded out while recording for long enough to leave a quiet spot.

For academicians or perfectionists, the recorded selections can be separated and stored on separate reels, so that each one may be filed according to composer, or they may be kept together and a cross-indexed file system set up so any number can be readily located.

Since a library of off-the-air recordings is presumably to be kept for some time, it is advisable that recordists observe the few simple precautions in storing tapes that will help to keep them in their original condition for as long as possible. These precautions were discussed in this department last April.

WOODCRAFTER

Continued from page 9

important safety measures to keep in mind is this: never have more than 1/8 in. of the blade exposed above the stock being cut. Should there be an accident, the injury would be not more than 1/8 in. deep. Keeping the blade close to the work also reduces the amount of dust raised and, in the event of a slight twist in the work, minimizes the force of kickback of the stock.

The *tilt handle* or *tilt lever* is usually situated on the left side of the machine and tilts the blade to the desired angle as indicated by the built-in tilting scale. A locking device prevents the blade from shifting its angle.

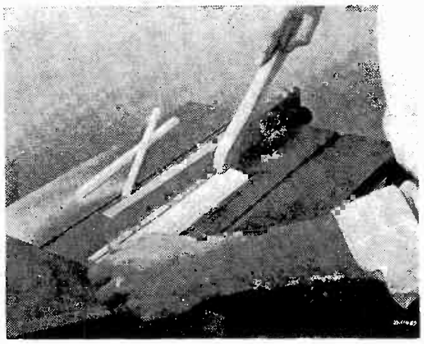
The *fence*, or *ripping fence* as it is commonly called, extends from the front of the saw table to the back and its distance from the blade can be adjusted to accommodate the work. It acts as a guide when cutting wood with the grain. To set the fence at the correct position, measure the distance from the fence to the edge of the blade tooth facing toward the fence. Lock the fence and begin the ripping operation.

The *miter gauge* is a device which slides the depth of the saw table in either of two grooves, guiding the work squarely or at an angle through the blade in crosscutting. The miter gauge should be checked with a try square for correct setting of the gauge pointer.

When you purchase a circular saw, the many operations which that particular machine can perform are described and pictured in literature prepared by the manufacturer. Absorb the basic directions thoroughly before turning on the switch for the first time. Previously we listed general safety practices for all power tools. Now let's be specific about circular saws.

Ripping: Never stand directly behind the saw blade, always a bit to one side. In fact, never let anyone watching you stand directly behind the blade or a kickback of the work might cause injury. (I know of one instance when the saw was powerful enough to throw back a piece of wood, which pierced the panel of a heavy door and traveled 100 ft. beyond!)

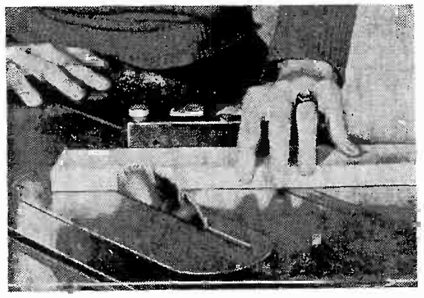
COURTESY ROCKWELL MFG. CO., DELTA POWER TOOL DIVISION



A push stick used to rip thin strips.

For ripping narrow strips of wood under 3 in. wide, use a stick, not your hand, to push the work through the saw; if there are less than 3 in. between the ripping fence and the blade, keep your fingers away. In fact, even with more clearance never use your whole hand to push the work. Let your hand straddle the ripping fence so that three fingers push the work while the remaining two slide along the other side of the fence. They are the anchor fingers to prevent the whole hand from being pulled into the saw should anything go wrong.

COURTESY ROCKWELL MFG. CO., DELTA POWER TOOL DIVISION



Correct way to operate a miter gauge.

In feeding stock into the blade always apply pressure against the fence. Don't force the wood into the saw; use a smooth, consistent pace. After the stock passes through the blade, let it fall to the front of the machine.

Long boards, 6 ft. or over, are difficult and dangerous to rip without assistance from a helper to support the other end. A roller support of the same height as the saw table can be placed in front of the machine to receive the work and eliminate the need for another person.

Crosscutting: The edge of the work which rests against the miter gauge must always be straight to eliminate twisting and kicking back as the saw blade strikes it.

When feeding stock into the blade, hold the work firmly against the miter gauge and be certain that your hand is well out of range of the blade.

Never pass the work through the blade with the edge of the wood held against the miter gauge while the end is pushing against the ripping fence. In other words, never use the fence itself as a length gauge because the danger of kickback is great. A safe and practical length gauge can be improvised simply by clamping a short block to the inside of the fence at the front of the table. This keeps the fence itself back from the work as it travels into the blade.

The circular saw is a fascinating tool to operate because it can perform so many functions in the home workshop. Here's the secret of making it work accurately for you in cabinetry where it is essential that multiple parts be cut identically: once you have set up the machine make *all* the cuts for which that setup is intended before breaking it down and proceeding with the next setup. It's practically impossible for the amateur craftsman to duplicate the precise measurements of a setup.

COURTESY ROCKWELL MFG. CO., DELTA POWER TOOL DIVISION

Roller support being used to cut long boards. The cut is easier and safer to make.

SOUND SERVICING

Continued from page 11

invariably are better in "low-boy" cabinets, preventing a direct blast of sound and producing a certain amount of absorption by the living-room carpet. You may want to experiment quite a bit with speaker positioning to get rid of the blast of "highs" that all too many manufacturers seem to think is a requirement for high fidelity.

GROUNDED EAR

Continued from page 5

market in commercial form and has several features which deserve comment. For one thing, each cartridge comes with a serial number and an individual test

chart of frequency response. The chart is not in conventional form, though, and will be almost meaningless to the average purchaser. Apparently it gives response at several discrete frequencies within its range, as plotted with a pen recorder. It would be more satisfying if the booklet to which the chart is attached indicated just what frequencies are involved. But as it stands it is evidence of fairly rigid quality control and, in the event the cartridge is returned for service, the manufacturer will have a check against the original performance.

Second, the stylus replacement method is quite possibly the simplest and most foolproof yet. Any child can change the stylus assembly without tools. Furthermore, the stylus is mounted and positioned in such a way that the accumulated lint and dust does not, so far as I can see, close any gap or restrict needle motion significantly. Also, brushing off the dust and lint can be done without danger of brushing away damping materials.

Finally, the mount is readily adaptable to most arms and, since the cartridge itself can be removed with a flick of the finger, it can be mounted easily even in the most cramped space. For that matter, the electrical performance is also exceptionally good.

HEATH PREAMP

Continued from page 29

1 and 2 volts output, respectively. With the volume control at ¾ rotation, distortion approaches insignificance: 0.13% and 0.33%, respectively, for 1 and 2 volts output. But with maximum volume setting the corresponding figures are .06% and 0.14%. It may as well be zero, as far as the ear is concerned.

The signal level fed into the unit is unimportant here, because the incoming

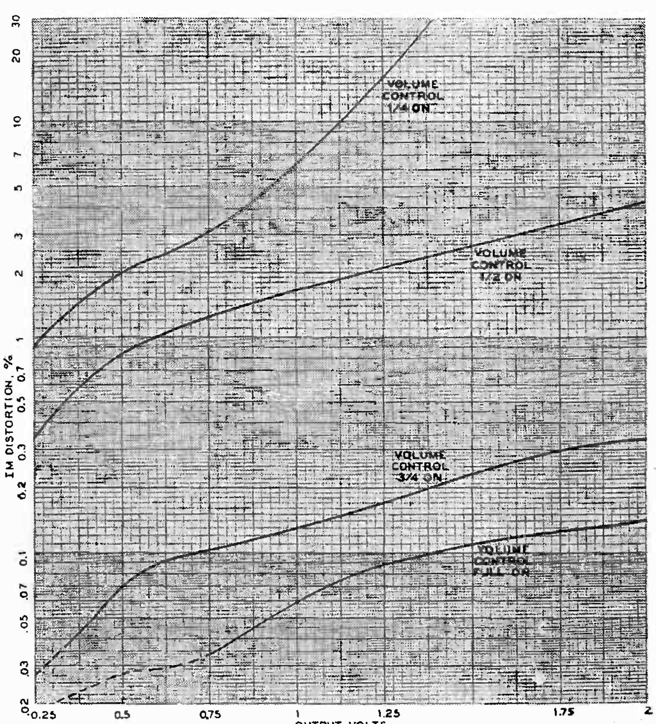


Fig. 10. Distortion on high-level inputs, as a function of output voltage, for four positions of volume control.

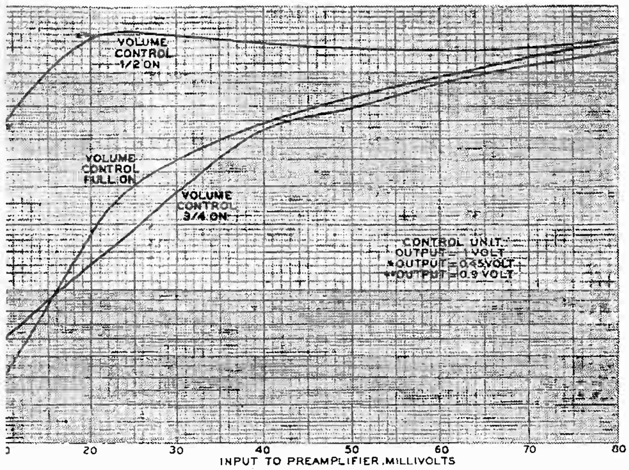


Fig. 11. Distortion on low-level inputs, as a function of input voltage, for three positions of volume control.

signal is piped directly to an input level control. If this control is adjusted so that the main volume control must be turned to ¾ full rotation or beyond in order to obtain maximum usable loudness, you can be assured that the distortion will be insignificant. This doesn't necessarily follow for low-level inputs, though, since incoming signals are amplified by the preamplifier stage before their levels can be adjusted by level controls. Fig. 11 shows the performance on low-level inputs. Here IM distortion is plotted as a function of input signal level for three positions of the main volume control; output was held at 1 volt by adjustment of the back-panel level control, except for the points noted on the chart (for which it was impossible to obtain 1 volt output at that volume control setting and for those input levels).

The upper curve, that for ½ volume control rotation, shows that for any input level at which it is possible to obtain 1 volt output, distortion is above 1%. It is obvious that this distortion originates primarily in the control section at input levels up to about 40 mv. It is also plain that some distortion cancellation occurs; as the input level increases about 20 mv, distortion in the preamplifier section cancels some of the control-section distortion, indicated by decreasing total distortion. This is verified again by the slightly lower distortion obtained (at any input level above 16 mv) with the volume control at ¾ rotation compared to that with full-on volume control setting. With any volume control setting of ¾ rotation or more, the distortion for 1 volt output is less than 0.5% up to 35 mv input. Up to 16 mv input, the same output can be obtained with less than 0.1%! The advantage of low-output phono cartridges, incidentally, can be seen also — they don't drive the preamplifier stage so hard.

To sum up, then: keep all input level controls down far enough that the main volume control has to be turned all the way up to obtain all the volume you can use, and you'll have a preamp-control that compares favorably with the best — for $20 and a weekend of spare time.

WHY EQUALIZE

Continued from page 25

back-head response compensates the recorded treble droop CD. Bass-boost equalization FE is then required in the playback amplifier to compensate for the portion AE of playback-head response.

In sum, when playing an NARTB tape, the playback amplifier must contain *electrical* equalization corresponding to the bass-boost Curve FE in Fig. 8. It should be noted that Curve FE tapers off over the portion FI. This is because the recorded tape already contains a slight amount of bass boost, as shown by Curve HG.

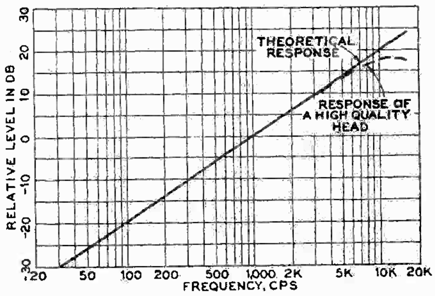


Fig. 7. Idealized head pickup response.

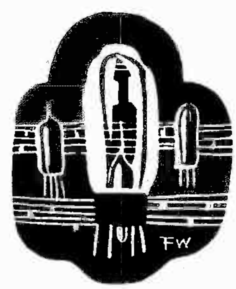
Returning to Fig. 6, it will be recalled that the treble droop of the recorded signal is the net result of losses that take place in recording, and of treble emphasis, the latter carried about as far as it can be without incurring distortion. Since treble boost brings the risk of distortion, it might seem advisable to lessen this risk by reducing treble boost, so that the curve in Fig. 6 would begin to decline at a lower frequency. Correspondingly, more treble boost would be required in playback, achieved,

Continued on page 40

Abbreviations

Following is a list of terms commonly used in this magazine, and their abbreviations. The list is arranged in alphabetical order.

alternating current	AC
ampere, amperes	amp, amps
amplitude modulation	AM
audio frequency	AF
automatic frequency control	AFC
automatic gain control	AGC
automatic volume control	AVC
capacitance	C
cathode ray tube	CRT
characteristic impedance	Z_0
current	I
cycles per second	cps
decibel	db
decibels referred to 1 milliwatt	dbm
decibels referred to 1 volt	dbv
decibels referred to 1 watt	dbw
direct current	DC
foot, feet	ft.
frequency	f
frequency modulation	FM
henry	h
high frequency	HF
impedance	Z
inch, inches	in.
inches per second	ips
inductance	L
inductance-capacitance	LC
intermediate frequency	IF
intermodulation	IM



kilocycles (thousands of cycles) per second	Kc
kilohms (thousands of ohms)	K
kilovolts (thousands of volts)	KV
kilowatts (thousands of watts)	KW
low frequency	LF
medium frequency	MF
megacycles (millions of cycles) per second	Mc
megohms (millions of ohms)	MΩ
microampere (millionth of an ampere)	μa
microfarad (millionth of a farad)	μfd
microhenry (millionth of a henry)	μh
micromicrofarad	μμfd
microvolt (millionth of a volt)	μv
microwatt (millionth of a watt)	μw
milliampere (thousandth of an ampere)	ma
millihenry (thousandth of a henry)	mh
millivolt (thousandth of a volt)	mv
milliwatt (thousandth of a watt)	mw
ohm	Ω
permanent magnet	PM
potentiometer	pot
radio frequency	RF
resistance	R
resistance-capacitance	RC
resistance-inductance	RL
revolutions per minute	rpm
root-mean-square; effective value	RMS
synchronous, synchronizing	sync
television	TV
ultra high frequency (radio)	UHF
vacuum-tube voltmeter (multipurpose)	VTVM
vacuum-tube voltmeter for AC measurements only	AC VTVM
variable reluctance	VR
very high frequency (radio)	VHF
volt	v
volt-ampere	va
voltage, or potential difference	E
volts, center-tapped	vct
watt	w

AUDIOCRAFT
Sound Sales Directory

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

KEY TO PRODUCTS HANDLED

1 Audio system components
2 Speakers and enclosures
3 Records and record accessories
4 Tape recorders
5 Pre-recorded tape
6 Radio hardware
7 Tools, wood
8 Audio parts
9 Microphones
10 Books
11 Test equipment

CALIFORNIA

Culver City
Bar-Shel's Hi-Fi Supply
5512 S. Sepulveda Blvd.
EX. 8-6747 1, 2, 4-6, 8-11

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

Los Angeles
Crenshaw Hi-Fi Center
107 Santa Barbara Plaza
Axminster 3-8201 1-5, 9

Menlo Park
Science in Sound
777 Santa Cruz Ave.
DA. 2-1611 1-11

Santa Barbara
Pacific Audio Supply
2919 De La Vina St.
Woodland 5-6422 1-5, 9

CONNECTICUT

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

DELAWARE

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

FLORIDA

Miami
High Fidelity Assoc.
3888 Biscayne Blvd.
82-8401 1, 2, 4, 5, 8-10

GEORGIA

Atlanta
High Fidelity Sight & Sound Sys.
606 Peachtree St., N. E.
Vernon 6534 1-10

ILLINOIS

Champaign
The New Sound
35 E. Springfield Ave.
6-9119 1-5, 8-11

Chicago
Allied Radio Corp.
100 N. Western Ave.
Haymarket 1-6800 1-11

Allied Radio Corp.
2025 W. 95th St.
Beverly 8-1067 1-11

ILLINOIS — *Chicago (continued)*
Musicraft
48 East Oak St. 1-11

Voice and Vision, Inc.
Rush and Walton Place
Whitehall 3-1166 1-6, 8-10

Evanston
Allied Radio Corp.
602 Davis Street
Davis 8-8822 1-11

LOUISIANA

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

MASSACHUSETTS

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

MICHIGAN

Detroit
Hi-Fi Studios
8300 Fenkell
DIamond 1-0894 1-6, 8-10

MISSISSIPPI

Pass Christian
The Music Box
121 Davis Ave.
185 or 301W 1-5, 8, 9

MISSOURI

Kansas City
David Beatty Hi Fi Sound & TV
1616 West 43rd (Westport Rd.)
Jefferson 1-3110 1-10

NEW YORK

Albany
Hi-Fidelity Center, Inc.
324 Central Avenue
3-1167 1-5, 8-10

Buffalo
Frontier Electronics, Inc.
1507 Main Street
GA. 5727 1-5, 8-10

New York City
Arrow Audio Center
65 Cortlandt Street
Digby 9-4730 1, 2, 4-6, 8-11

Grand Central Radio, Inc.
124 East 44th St. at Lexington Ave.
MUrray Hill 2-3869-70 1-6, 8-11

Yonkers
Westlab Electronics, Inc.
2475 Central Avenue
SP. 9-6400 1-6, 8-10

OHIO

Dayton
Custom Electronics, Inc.
1000 S. Main Street
AD. 3158 1-11

PENNSYLVANIA

Allentown
Baker Communications
9 South 12th Street
Hemloch 3-3326 1-11

PENNSYLVANIA *(continued)*

Bethlehem
Audio Laboratories, Inc.
808 Mohican St.
UNiversity 7-3909 1-5, 9

Philadelphia
Almo Radio Co.
412 North Sixth St.
WAlnut 2-5153 1-4, 6-11

Danby Radio Corp.
19 South 21st Street
Rittenhouse 6-5686 1-5, 8-10

Ten Cate Associates
6128 Morton Street
Germantown 8-5448 1-5, 8-10

SOUTH CAROLINA

Columbia
HI-FI Sound & Records Co.
621-23 Harden St.
6-3538 1-6, 8-10

TENNESSEE

Kingsport
Radio Electric Supply Co.
245 East Market St.
Cl. 5-1471; 5-2651 1, 2, 4, 6, 8-11

TEXAS

San Antonio
Bill Case — Records & Sound
4206 Broadway
TAylor 2-1341 1-5, 8-11

VERMONT

Rutland
Fleetwood Sound Studio
285 S. Main Street (Rt. 7)
Prospect 3-2312 1-5, 8, 9

VIRGINIA

Arlington
Audio Services
3131 N. 10th Street
Jackson 7-2552 1, 2, 4, 5, 8, 9

WASHINGTON

Seattle
High Fidelity Headquarters
603 Broadway North
CApitol 2266 1-4, 10

WEST VIRGINIA

Huntington
Singer Electronics
1555 Third Avenue
3-7439 1-5, 8-10

WISCONSIN

Milwaukee
The Hi-Fi Center, Inc.
4236 N. Capitol Drive
Uptown 1-2113 1-5, 9, 10

CANADA

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

Toronto
Electro-Voice Sound Systems
141 Dundas Street, W.
EM. 8-7750-8301-8302 1-11

WHY EQUALIZE

Continued from page 37

as previously explained, by means of the head's rising response characteristic. Such a course, although it could be followed, would have the disadvantage of reducing the tape recorder's signal-to-noise ratio in two ways: 1) less treble boost in recording means a smaller

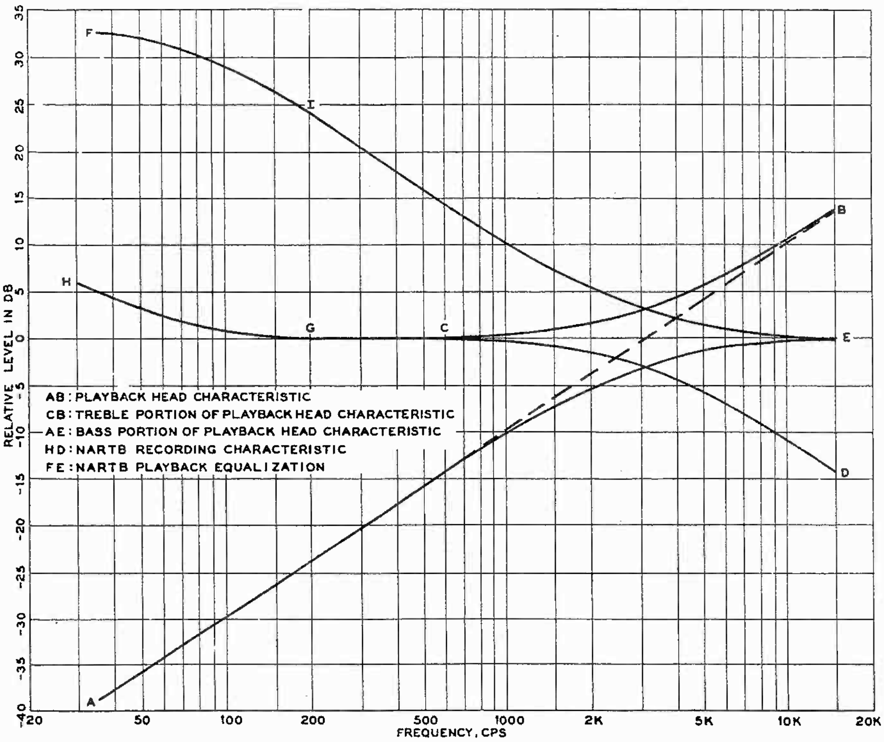


Fig. 8. How flat over-all response is achieved with optimum signal-to-noise ratio.

amount of signal recorded on the tape; 2) more treble boost in playback means greater accentuation of tape hiss, which can be an important component of noise. All told, therefore, the NARTB recording characteristic represents what is considered to be a good compromise between low distortion and high signal-to-noise ratio.

The reason for confining electrical treble boost (we are distinguishing treble boost circuits from the playback head's rising characteristic) to the recording process, according to the NARTB standard, has been partly suggested: boost in playback would emphasize tape hiss. Furthermore, noise generated in tubes and resistors would be emphasized to an objectionable degree because the playback amplifier is a very high-gain affair.

As can be seen from Fig. 6 and Curve FE in Fig. 8, bass boost takes place essentially in playback under the NARTB standard. The reason is that an appreciable amount of bass boost in the recording amplifier would overload the tape, especially since tapes overload somewhat more easily at the very lowest frequencies. The small amount of bass boost used in recording, shown in Fig. 6, is considered to be about the safe limit. This recording boost reduces the playback boost requirement somewhat, and correspondingly decreases reproduction of hum and motor rumble.

ADVERTISING INDEX

TRADERS' 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.

FUN-FILLED recorded tapes! "Sampler" and lists, 99c. Complete "Radio Announcing" course on tape, only $10.00. Free details. Stone, Lunenburg 11, Mass.

D42 CONDENSER MICROPHONE ± 1 DB 0-10 KC, Low distortion, high sensitivity, rugged. $20.00. Write for data. C. Tendick, 900 Glenway Dr., Inglewood, Calif.