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VOLUME 1

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AUDIO

. . . the *original* magazine about high fidelity

in this issue:

AUDIOCLINIC by Joseph Giovanelli

EQUIPMENT PROFILES by C. G. McProud, Editor of **AUDIO**

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Foreword

In answer to the hundreds of letters requesting it, here is the first volume of a new series of books which AUDIO, through its publisher RADIO MAGAZINES, INC., will publish as a compendium of audio knowledge to be used as a reference and guide.

In Volume I, we have compiled *the-best-of* the AUDIOCLINIC edited by Joseph Giovanelli, noted audio engineer, authority and the *original* high fidelity *answer-man*. It is a compilation of the most important issues and facts about high fidelity, answering some of the most perplexing high fidelity problems in all phases of audio techniques.

Part two of Volume I is a compilation of EQUIPMENT PROFILES that have appeared in AUDIO from the Fall of 1957 to the present. Presented in simple, matter-of-fact style, it is a profile analysis of high fidelity components in action...definitely not test reports. It is a factual and thorough discussion of what makes a high fidelity component tick – what it does – how it does it. To quote an AUDIO reader... "It is a valuable reference for the high fidelity shopper."

In time we will publish other issues of "the best of AUDIO", presenting in one convenient volume the best editorial material on the subject of high fidelity. To all of you who have written requesting this book, we dedicate this first volume.

C. G. MCPROUD, *Editor*

Mineola, New York
June, 1959

CONTENTS

Frequency Response

Q. Is it possible to sweep a high fidelity preamplifier or amplifier and observe the over-all response curve as is done in video alignment? If so, how? Also, how can I determine the frequency response of a loudspeaker? Robert A. Poltzer, Chicago, Ill.

A. Yes, methods for rapidly sweeping the audio spectrum are often employed. The audio generator is made to sweep the spectrum and is, of course, fed into the device to be checked out. The output of the device is connected to a scope for direct display, or to an assembly which moves a pen over a moving drum in accordance with the dips and rises in response. By this means, a permanent graph of the response of the equipment is obtained.

Measuring the response of a loudspeaker is quite complicated because of at least three variables which you will encounter besides that of the speaker itself. Were it not for these, the method would be quite simple. Feed a series of audio-frequency tones into the speaker. Place a microphone near the speaker to pick up these tones. The output of the microphone feeds the measuring device.

The first question which arises is: how flat is the response of the microphone? In order to be sure, you have two courses open to you. One is to use a calibrated microphone, especially designed for this type of work. The second is to send your own microphone to the Bureau of Standards to be calibrated. Then, superimpose the graph of the microphone's response over that of the plotted response of the speaker in order to obtain the speaker's true response.

Another problem is that of the room in which the measurements are made. Any resonances, antiresonances or reflections present within the room will greatly influence the response curve. It is necessary to make these measurements in a room designed especially for such work. Such a room is known as a space room, or anechoic chamber, because of its complete freedom from reverberations. Most of us do not have such rooms available, but we can approximate the conditions found in them very closely by making the measurements out-of-doors. A rooftop is a good spot, but it may be resonant at several frequencies, and the effect of this will to be given erroneous response readings. An unpaved area considerably removed from buildings or trees is probably the best location, but here you can run into the danger that ambient noise may be picked up by the microphone.

The last variable which may be encountered is that of the angle at which the soundwaves strike the diaphragm. It is undesirable to measure the frequency dispersion of the unit, rather than its frequency response.

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High Fidelity

Q. I am a new subscriber to Audio, *and wanted immediately to avail myself of your offer to answer any and all questions, regardless of their suitability for use in the column. Since I am just beginning my hobby, I should like to know just what we are striving for. What is high fidelity? Harvey Bond, Hayes City, Iowa*

A. Fidelity indicates faithfulness to something. In this instance, we are interested in the degree of faithfulness with which reproduced sound (primarily in the home living room) resembles the original performance of that music or other material. What we wish to do is to make that resemblance as close as possible. High fidelity does not mean high volume of sound, and it does not connote an exaggerated emphasis of either the bass or the treble end of the audio spectrum. High sound volume is appropriate only when the original performance contained loud sounds. As has already been said, we want to make the reproduced sounds as nearly like those heard at their original performance as possible. This implies that we have not yet arrived at a perfect result; probably we shall never do so. We can improve our results, however, and that is why we work to get that last cycle in the bass or treble, that fraction of a per cent less distortion, or that extra db or two increase in signal-to-noise ratio.

One problem which confronts us, therefore, in defining high fidelity is this: Just how close to the ideal must our reproducer perform before we term it high fidelity? No one has as yet set the standard for this, and for that reason many packaged units are sold with letters emblazoned on their labels "HIFI"; to those of us who have heard truly good sound reproduction, it is hard to accept the sound produced by a single 4-inch speaker as really high fidelity sound.

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High Fidelity Equipment and Dampness

Q. *I have just purchased an Altec Lansing 820C speaker for my seashore home. Though I have the house on low heat all the time, dampness is ever present at the shore, and I have been told that dampness will affect the paper cones of the speakers or possibly cause the speakers themselves to rust.*

I would welcome your best opinion as to whether this dampness would also have some adverse effect upon amplifiers, preamplifiers, etc. What means, chemical or otherwise, might be employed to overcome this? John Sabritt, Philadelphia, Pa.

A. You are quite correct in your concern as to what dampness can do to high fidelity equipment. Of course, much depends upon the degree to which the equipment is exposed. Extremely damp, salty air can easily cause cones to go off center, and can corrode many of the parts of your equipment, leading to a breakdown of many of the capacitors and to freezing of the controls.

Fortunately, there are things which can be done to overcome this problem partially. One thing which can be done is to place silica gel in the boxes or cabinets in which the equipment is contained. In addition, you could include in each cabinet to be protected, a device known as a Dampchaser. This device is used extensively by piano tuners and manufacturers to keep pianos dry and at a constant temperature. These devices are also used by many manufacturers of electronic organs to protect the chassis of these instruments from the ravages of dampness. The Dampchaser is nothing more than a heating element. Its purpose is to raise the temperature of the device being protected two to four degrees above the surrounding temperature; this simple act will reduce condensation. These units are available in several sizes, and I would recommend that you use the largest possible size.

Once the equipment being protected is turned on, its own operating temperature will be sufficient to avoid condensation. Because of this, the Dampchaser need not operate at this time. This can be accomplished quite easily. Simply connect the linecord of the Dampchaser across the terminals of the switch of your system. When the device is turned on, the Dampchaser is automatically shorted out of the circuit. When the equipment is turned off, current can flow through the device. Since the power consumed is very small compared to the device being protected, most of the voltage will be developed across the Dampchaser, and almost none across the primary of the power transformer.

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Hum and Oscillation in Home Music systems

Q. *My sound system consists of a Miracord XA100, Miratwin cartridge, EICO HF61K preamplifier, EICO HF60K amplifier, and an AR-1 and AR-2 speaker systems. With the above equipment there are two difficulties which I have been unable to locate and correct.*

1. The system will oscillate at a low frequency when volume is fairly high and the input signal consists of a low frequency. It may be made to occur in the runoff grooves of some recordings. At times it may be caused by a tap on the turntable spindle while the arm and cartridge are in the operating position. From material appearing in AUDIO, *I understand that the feedback in amplifiers can be quite critical, and when improperly adjusted because of parts failure, can cause the oscillation of which I spoke. Could this be the cause? Of course, the trouble may be coming from the preamplifier.*

2. During the first half hour of operation, there is no more than a trace of 60-cps hum heard in the loudspeakers. After this period the hum becomes quite pronounced. After the equipment has been off for half an hour or more, it can be operated for a half hour before the hum reappears. The actual timing is only approximate. Tubes check satisfactory. Any suggestions you can furnish will be greatly appreciated. W. H. Focht, Tipp City, Ohio.

A. 1. I recommend that you check the electrolytic capacitors. If these are low in capacitance, they will present a means of common coupling to all stages. Before looking into the feedback loop, first determine whether noise is generated within the preamplifier or power amplifier. Disconnect the preamplifier and feed a signal directly into the amplifier. If you cannot cause the oscillation, it is possible that the preamplifier is involved. If the trouble is in the power amplifier, and you find that the trouble is in the feedback loop, you will probably find that the feedback resistor has changed value. A further discussion of the problems of checking feedback loops can be found later in this column.

The oscillation may be caused by acoustic feedback, rather than from any electrical failing of the preamplifier or power amplifier. It may be of two kinds. One is the result of the turntable being vibrated by the loudspeaker. Those vibrations are passed on to the amplifier and fed back to the speaker which then vibrates the turntable again, thereby sustaining oscillation. The other type of feedback is similar, but in it, the vibrations are picked up by a microphonic tube instead of by the cartridge. The elements in such a tube are free to move slightly, changing the gain of the tube. This causes a noise to be heard from your loudspeakers.

2. The 60-cps hum is probably caused by a heater-cathode leak in one of the tubes in your equipment. This leak will not show up on a tube tester because it does not exist until the tube has been in operation for some time.

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Impedance Measurement

Q. *How may I determine the impedance of a device such as a phonograph pickup or microphone? Ira Jamieson, Rockville Centre, New York*

A. *Figure* 1 shows the normal wiring of such a device. R_L is shown in parallel with the device, and its value is so chosen that it will be equal to the impedance of

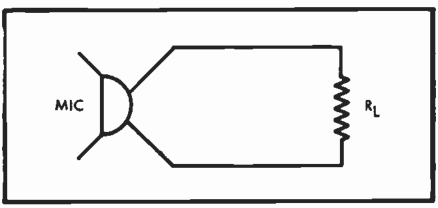

Fig. 1

the device. In many cases, where the impedance is low, a transformer is substituted for the resistor. To determine the impedance of the device, feed a signal into it. In the case of a microphone, this is accomplished by placing it near a loudspeaker and feeding this speaker in turn with a tone whose frequency is approximately 400 cps. If the device is a phonograph cartridge, use a frequency record. Connect the output terminals of the device under test to the input terminals of an a.c. voltmeter whose impedance is at least ten times that of the device under study. A VTVM with an input impedance is a few megohms and a sensitivity of around 1 millivolt can work well with most devices. (If the unit being measured is a crystal microphone, its probable impedance is 5 megohms, so that a VTVM with an input impedance of at least 50 megohms would be needed and this is not usually available.) *Figure* 2

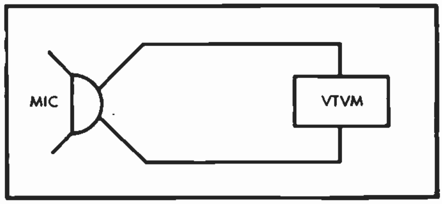

Fig. 2

shows the equipment under test wired to the measuring instrument. Note the voltage read under these conditions. Now connect a variable resistance across the device as shown in *Fig.* 3. Adjust this resistance until the output has dropped 6 db (which occurs when the voltage reading is half that of the original, or unloaded, voltage). If, regardless of the setting of the resistor, the voltage does not rise to this value, the maximum resistance is too small and a larger one must be substituted. If the adjustment is critical and falls close to the minimum resistance of the element, it would be better to substitute a smaller unit for the one originally used. At any rate, once you have arrived at the point where the voltage has been dropped 6 db, disconnect the resistor from the circuit

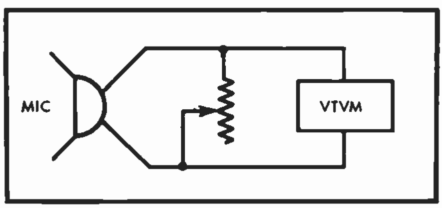

Fig. 3

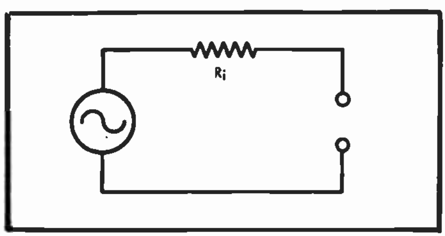

Fig. 4

and measure its value with an ohmmeter. The ohmic value of this resistor is equal to the impedance of the device being tested.

The impedance of amplifiers may be measured by similar means. A signal is fed into the amplifier and the unloaded secondary winding of the output transformer is connected to the indicating meter.

This time, however, a vacuum-tube-voltmeter need not be used. The standard 20,000-ohms-per-volt movement will work very satisfactorily, as will an output meter whose impedance is as low as 100 ohms. Be sure that you do not apply too much signal to the input terminals of the amplifier because the voltage across the terminals of the output transformer will rise to a level which might be sufficient to cause arcing within, and this would probably ruin the transformer. The variable resistor is adjusted as before. The impedance read this time will be the source impedance of the amplifier. This is something quite different from the impedance into which the amplifier is designed to work. An amplifier might have a source impedance of 0.5 ohm, but the impedance into which it is intended to work is 8 ohms.

The question naturally arises as to why putting a resistor in parallel with a cartridge or amplifier can yield the impedance. You place almost any number of appliances in parallel across a 117-volt line without causing much drop across the line. All that is proved by this logic is that the impedance of the line is very low. Do not use the foregoing means to measure the line's impedance, for the least that can happen is that you will blow the house fuse in the attempt, and cause serious overheating of the wiring. The answer to this lies in our reconsidering the internal structure of the pickup. The resistor used to measure the impedance of the pickup is really in series with it and not in parallel with it. The pickup may be considered as being composed of a generator of zero resistance in series with the internal impedance of the pickup. This is shown in *Fig.* 4. When the load resistor, or test resistor, is placed across the pickup, we are actually placing it in series with the pickup's internal resistance or, more accurately, its internal impedance, R_i in *Fig.* 4. *Figure* 5 shows that the generator, the pickup's

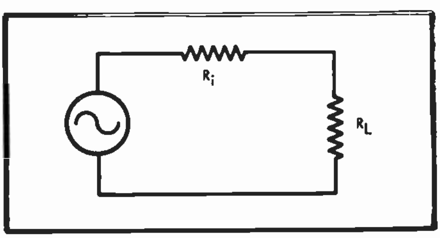

Fig. 5

internal impedance, R_i, and the external load or test resistance, R_L, are in series. R_L and R_i form a voltage divider. When R_L is large with respect to R_i, a larger voltage will be developed across it than R_i. In fact, the voltages will divide in accordance with the ratios of the values of the resistances. Therefore, when the two resistances are equal in value, the voltage drops across them will be equal. The generator is supplying a constant voltage to the two resistors. When R_L is infinite and R_i some finite value of resistance, all the voltage available will develop across R_L. As the value of R_L is reduced, more voltage is dropped across R_i and less across R_L. When the values of R_i and R_L are equal, equal voltage drops will appear across the two resistances. Because the voltage of the generator has remained constant, the voltage across R_L is half as much as it had been when R_L was infinite. (When the test instrument has an impedance at least ten times that of the device being tested, the impedance of the meter may be considered infinite, and therefore will have negligible effect on the results of the test.

The reason for measuring impedance in this manner, rather than directly with an ohmmeter is that inductance and/or capacitance in the device to be measured cause it to have an impendance value different from the *resistance* value that which would be shown on an ohmmeter.

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Transformer Impedance

Q. I was recently given what seems to be a fine output transformer, but I don't know its impedance. The wires are divided into two groups, the first of which contains those colored red, blue, and reddish yellow, while the second contains those colored black, brown, orange, and yellow. The first group is evidently the primary, and the second, the secondary, but how will I proceed from here? Al Kerper, Brooklyn, N. Y.

A. My first recommendation would be to locate the model number and the name of the manufacturer of the transformer, and then consult his catalogue for the desired information. If, for any reason, you are unable to follow this course, proceed as follows:

Assume first that you have correctly identified the groups of wires. Red goes to one plate, blue to the other, and reddish yellow to B plus. (proper phasing of the primary leads can be found only by trial & error. If the phase is reversed, the amplifier will oscillate. Black is probably the common on the secondary winding. It is probable that the brown is the 4-ohm tap, orange the 8-ohm tap, and yellow the 16-ohm tap. However, to be certain, measure the resistance of each lead with respect to black, or common, arrange the leads in order of ascending resistance values, and they will be in order of ascending impendance values, although the resistances do not equal the impedances. You will find that the lowest resistance you will come across will be less than one ohm, whereas the impedance represented by this resistance is 4 ohms.

You now have the probable impedances of the secondary and next you must find the impedance of the primary. This is done by connecting a resistor of appropriate value across the secondary, feeding in a signal from an audio oscillator at a given voltage, and then noting the voltage appearing across the primary. The square of the voltage ratio between the signal fed in and the voltage appearing across the primary gives the impedance ratio of the two windings.

Illustration: Start with the 8-ohm tap. Connect an 8-ohm resistor from this tap to common. Connect your audio generator across this resistor and feed the secondary with 1 volt of signal of approximately 400 cps. Measure the voltage appearing between the red and blue leads (primary). Be sure to use a fairly sensitive a.c. voltmeter for this purpose, so as not to load down the primary circuit. Let us assume that you get a reading of 30 volts. Since the ratio of the voltage fed in to that appearing across the primary is 30:1, the turns ratio is also 30:1. The impedance ratio is equal to the *square* of the turns ratio, so we find that the impedance of the primary is 30^2, or 900 times that of the secondary. Since the impedance of the secondary is 8 ohms, the primary impedance must be 7200 ohms. This primary impedance is correct only when the secondary is terminated in an 8-ohm load. Within limits, the transformer can be used to match a range of impedances. The only thing which is really constant is the turns ratio. Do not confuse the impedance of the transformer with that of the internal impedance of the amplifier. This latter is a function of the amount of negative feedback applied.

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Capacitance and resistance

Q. Please give me the proper method for calculating the results of (1) wiring two or more resistors in parallel, (2) wiring two or more resistors in series, (3) wiring two or more capacitors in parallel, and (4) wiring two or more capacitors in series. Boyd H. Redner, Battle Creek, Mich.

A. The sum of two resistors wired in parallel can be found by the following formula: Total resistance equals the product of the resistances divided by their sum. This formula applies to capacitors wired in series, also. Example: Two 5-ohm resistors are connected in parallel. What is the resultant resistance? The product of the two resistances is 25 ohms, which must be divided by their sum, 10 ohms. The resistance of this parallel combination is, therefore, 2.5 ohms. This formula does not hold where two or more resistances or capacitors are involved. In such instance, proceed as follows: Invert each of the resistance values, add the resulting fractions, and invert the result. This final inversion will give you the answer. This method may be used for any number of units, including two. Example: Three resistances having values of 3, 4, and 5 ohms are connected in parallel. What is the resistance of the network? Solution: First, invert the fractions, and obtain 1/3, 1/4 and 1/5. The least common denominator for these fractions is 60, so we must add 20/60, 15/60, and 12/60. This totals 47/60, which, when inverted, becomes 60/47, which equals approximately 1.27 ohms.

The total resistance of resistors wired in series or the total capacitance of capacitors wired in parallel is equal to the sum of the individual values. Example: Resistances of 5, 10, and 20 ohms are wired in series. What is the resistance of the network? Add 5, 10, and 20, and obtain 35 ohms.

Be sure that all resistances and capacitances are computed in the same values. Do not work with 1000 ohms and 1 megohm without converting both into ohms or into megohms. 1000 ohms equals 0.001 megohm, and 1 megohm equal 1,000,000 ohms. Don't add micromicrofarad with microfarad values. 1 micromicrofardad equals 1/1,000,000 of one microfarad.

* *3420 Newkirk Ave., Brooklyn 3, N.Y.*

Matching Impedances

Q. How can I go about matching impedance of a tuner and the output or input impedance of an amplifier or preamplifier? H. T. Sutcliffer, Redwood City, Cal.

A. In audio work, it is rarely necessary to know the exact input or output impedance of a piece of equipment, though that of the output stage of a power amplifier and of some low-impedance input stages is somewhat more critical. With straight RC circuitry, all that is necessary is for the impedance of a stage being supplied with signal to be at least twice that of the driver. I usually establish this ratio at between five and ten to one. Cathode followers are of low impedance usually, and can be easily fed into amplifiers of many times their impedance without the use of matching equipment. The actual impedance of a cathode follower depends upon the tube employed and upon the cathode resistor. The input impedance to a particular amplifier or amplifier stage is roughly that of the load into which the coupling capacitor works. For example, the output impedance of a discriminator of an FM tuner is approximately that of the resistance between cathode and ground of the diode from which audio is derived (about 100,000 ohms, usually). The coupling capacitor should be 0.02 µf. or larger. The output of the capacitor should be terminated in a stage whose input impedance is approximately 0.5 megohm.

The output of a conventional grounded cathode a.c. amplifier is roughly that of the plate load resistor, and again, the value of the capacitor used to couple the signal to the next stage can be neglected.

Inductances in Series

Q. Given two similar unshielded chokes, salvaged from a TV power supply, which I want to add in series followed by a shunt capacitor to a choke-input power supply, is there a possibility of not getting combined inductance value unless they are connected in a certain way? If so, is there a simple method of determining the proper terminal arrangements?

A. When connecting chokes in series, you will find that their inductances will be additive, regardless of polarity of wiring. The only things which will change when the connections of such inductances are reversed will be the polarities of the back voltages. Even though the chokes are unshielded, coupling between them will not be sufficient to cause cancellations or reinforcements, as would be the case were the inductances wound on the same form.

Negative Feedback

Q. What is negative feedback, and what is its purpose? John Ivers, St. Louis, Mo.

A. By negative (inverse) feedback we mean that condition wherein a portion of the output is returned to a previous stage in phase opposition to the input signal. It may be looked upon as a peculiar kind of tug-of-war in which the output does nothing until the input receives a signal. Then the input voltage is opposed by what we call the feedback voltage. The main input signal voltage always wins the game, since it is impossible to reintroduce sufficient signal to the input circuit to cancel it completely.

Figure 1 shows a single audio-frequency

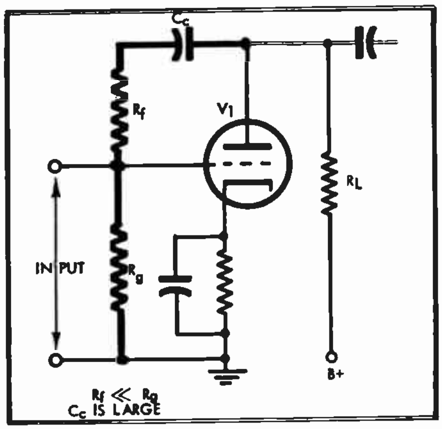

Fig. 1.

stage in which feedback is applied from plate to grid. There are two paths from the plate of V_1. One leads to a succeeding stage, while the other leads back to the input grid. The latter path delivers the signal to the input, and is known as the feedback loop.

From what has been said so far, it is obvious that a signal can be returned from output back to input with no difficulty. In order that we may call this signal *negative* feedback it must be in such a direction that it cancels some of the input signal. This circuit meets this requirement. At some portion of the input cycle the grid is made more positive with respect to ground. The tube draws more plate current, and the voltage dropped across the plate-load resistor, R_l, increases. This causes the plate to become less positive than it was with no signal applied to the grid. What I'm saying is that the plate and grid signals behave oppositely. Therefore, when the signal from the plate is fed back to the grid of the same stage, it is in such a direction as to cancel a portion of the input signal.

Not only is it important to be sure of the direction of the feedback, but it is also important to be sure of the *degree* of feedback. An examination of *Fig.* 2 will show that less feedback will be available to the input stage than was the case in *Fig.* 1. because the feedback resistor, R_F and the grid resistor, R_G form a voltage divider whose action is more severe in *Fig.* 2 than in *Fig.* 1.

Negative feedback is used for many different purposes, some of which will be given here.

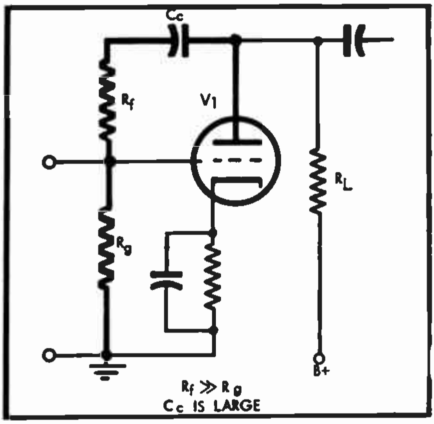

Fig. 2.

Negative feedback is used to correct frequency response. For example, if there is amplitude distortion present in an amplifier, the voltage produced at the distorting frequency will be greater (or less) than any other voltage fed to the feedback loop. (I am assuming, of course, that all voltages fed into the amplifier are equal.) Because the voltage of the distorting frequency at the output of the loop is greater (or less) than any other voltage at the output of the feedback loop, more of this voltage will be fed back to the input stage of the loop than at any other frequency. Thus, more (or less) of the distorting frequency signal will be canceled, and by this means response tends to level off. Notice I said "*tends* to level off." There is always present some error, much like that occurring in the governor mechanism of the old spring phonographs, wherein changes in spring tension were only partially compensated for by changes in the centrifugal force of the weights, and hence friction of the drum against its brake pads.

Another use for negative feedback is that of frequency compensation of the character found in tape and phonograph reproducers. The compensation is accomplished by making the feedback loop frequency sensitive. If what is needed is a circuit which will boost the lows, it is necessary to feed back more highs than lows, leaving the lows more or less unattenuated while the highs are considerably reduced in intensity. To accomplish this merely insert a small capacitor, C_C in series with the loop, as in *Fig.* 3. Because the reactance of the capacitor is high as compared to that of the feedback resistor, most of the lows are lost across the capacitor. The reactance to the highs, on the other hand, is small compared to that of the feedback resistor, and therefore, most of the voltage at these frequencies is available as feedback. If you desired to feed back more lows than highs, which you might in pre-emphasis networks, insert a small bypass capacitor in the loop. This will shunt the highs to ground, thereby bypassing the loop. This capacitor offers a low reactance path back to ground.

Sometimes the elements in these frequency compensating networks are made variable. When this is done we have a tone-control circuit most often referred to as the Baxandall type. Such variable networks are also the means by which many phono preamplifiers achieve the various record compensation curves.

Still another use for inverse feedback is that of making an amplifier more nearly linear. This is accomplished in a manner similar to that discussed in connection with frequency correction. Any instantaneous peak or dip in a wave shape is automatically compensated for by an increase or decrease in the amount of feedback applied. This doesn't mean that all frequencies fed into the equipment will be converted automatically into sinewaves. This would be highly undesirable, since much of the transient nature of program material consists of steep wavefronts. The feedback removes the discrepancy between the signal being

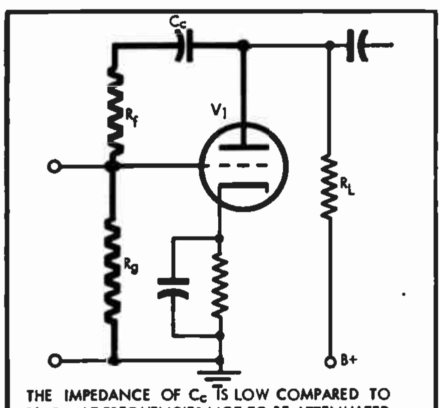

Fig. 3.

fed into the equipment and that which is reproduced in its output. This use of negative feedback will become increasingly important as more and more transistorized equipment comes on the market, because such amplifiers are operated Class B. Such operation is extremely nonlinear and can be overcome only by applying tremendous amounts of feedback.

Outside the field of sound reproduction feedback finds a wide variety of uses. Atomic reactors use feedback to control the rate at which neutrons are accelerated toward their targets. Such feedback is used in automation to reconcile differences between programmed work and the actual work produced. Various computer operations make use of the error signal from feedback as a means of arriving at the proper answer to the problem fed to the computer.

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Capacitive Reactance

Q. I have a motor and a medium-quality pickup arm. Recently I exchanged my crystal for a magnetic pickup. This will give me a wider frequency range, but at a lower voltage. On this basis, the chances are greater for distortion to occur in the form of hum fields produced by the motor. In more powerful amplifiers, some leads are shielded, the shield being grounded. However, if the shield does not have a fairly large diameter compared to that of the center conductor, some capacitive reactance will be present. This tends to attenuate the higher frequencies. It seemed to me that I should use shielded coaxial wire as lead-in for the magnetic cartridge to protect it from the magnetic field of the motor. Will the capacitive reactance tend to cancel the advantages of the shield?

A. Your statements about capacitive reactance of shielded cables are, of coure, true. However, if the length of such cables is kept as short as possible, the reactance at audio frequencies as compared to the impedance of the cartridge used is high. Therefore, the wide frequency range of such cartridges is not shunted, and can be passed on to your preamplifier. I cannot specify the exact length of cable which can be used without losses because so much depends upon the type of cable and upon the impedance of the cartridge used. Magnetic cartridges used in modern home music systems range in impedance from an ohm or two to around fifty thousand ohms. The higher the impedance, the shorter the length of cable which can be tolerated.

RMS

Q. What is RMS? Name withheld.

A. Measurements on the a.c. supply voltage used for home illumination shows it to be 115 volts, approximately. But this voltage is constantly varying from zero to a maximum value, back to zero, to an equal maximum of opposite sign, then back to zero, to begin the cycle over again. Since the maximum values of a.c. voltage are instantaneous, the *effective* voltage is less than this maximum. While the maximum voltage appearing across your house wiring system may be 150 volts, it is no more effective in doing work than an equivalent d.c. voltage of 115 volts. Most a.c. voltmeters are calibrated to indicate this effective value. It can be shown that this effective value is 0.707 of the maximum value (with a sine wave), while the maximum value is 1.414 times the effective value. This is arrived at in the following manner: As many instantaneous voltages along a cycle as practical are first squared, then added up, divided by the number of points involved, and the square root is extracted. It is from this process that we get the term RMS, *root mean square.* All we are doing is taking an average but, because of the sinusoidal nature of the alternating voltages, we must use squares and square roots as well as the standard means of taking an average.

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Two Amplifiers

Q. How can I connect two 40-watt amplifiers to drive one 4-ohm speaker system? Both are to be fed from the same preamplifier. Frank A. Cappi, Chicago, Illinois.

A. I do not advise connecting your two amplifiers to one speaker system unless you are sure that the speaker system can handle their combined power. One of your amplifiers has more than enough power output to overload the speaker system. The only value of so connecting your amplifiers would be that of obtaining slightly less distortion. Although the decrease in distortion could be measured by laboratory methods, it is doubtful that it could be detected aurally. Should you still wish to try this, proceed as follows:

1. Connect the output of the preamplifier directly to the input of each of the power amplifiers.
2. Connect together the common, or ground, terminals on the power amplifiers.
3. Join the two four-ohm taps. This will provide the necessary four ohms to operate the speaker.

By so connecting the equipment, more power output is obtained than could be had from a single amplifier. It should be remembered, however, that the internal impedance of the output circuit has been halved, which is the reason that the 4-ohm speaker had to be connected to what had been, for each unit, the 4-ohm takeoff.

Since the output of most preamplifiers is terminated in a cathode follower (having low output impedance), two or more high-impedance circuits may be directly bridged across its output terminals without significantly affecting the performance of the system. When connecting power amplifiers as described, be sure that the amplifiers are identical. Should one have one more stage than the other, the output signals would be 180 deg. out of phase, causing a cancellation rather than a reinforcement of the signal. Even if the amplifiers have the same number of stages, but are of different circuit design, there is likely to be enough phase difference between their output voltages so that at least partial cancellation will result.

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Low Gain of Power Amplifiers

Q. Thanks for your reply to my recent letter. I have another problem which has plagued me for some time now. Several years ago I built a 20-watt Ultra Linear amplifier using two 807's, two 6SN7's and a 5U4. It has served me faithfully for some time but it has developed troubles which I cannot locate. The gain has fallen off considerably, and new tubes did not help. Voltages all check normal. I replaced the filter capacitors because, with my input level control rotated all the way to the left, motorboating was audible. The replacement of these capacitors did not help. I would appreciate hearing from you concerning this matter with any advice you can give. James O. Valestin, St. Louis, Mo.

A. The first thing to check is the cathode bypassing. Failure of such a component will result in a reduction in gain and perhaps some instability. A drop in gain accompanied by an even greater reduction in bass would be further evidence that a cathode bypass capacitor is open.

Sometimes the amplifier will behave abnormally if there is an open grid resistor present. Under this condition, the grid is charged excessively with electrons from the cathode of the tube, thereby cutting itself off.

Another possible source of the loss in gain is that of defective coupling capacitors.

Still another source of trouble can be a cold-solder joint. Such joints seem good at the time they are made and indeed, they may work properly for some time, but ultimately some resin will penetrate among the various leads making up the connection and this will cause the resistance to rise, sometimes to infinity.

Another possibility is that one half of the output transformer has opened. This can lead to both loss of gain and to instability, especially when the output stage derives its bias through a dropping resistor in the cathode circuit.

Another possibility is that the feedback-loop resistor has changed value. If it has become smaller, more and more voltage will be fed back from the output stage, thereby reducing the gain of the amplifier. Further, excessive feedback can cause instability because of shifts in phase of some of the components, especially the output transformer. Although there are always phase shifts, they are not always great enough in their effects until the feedback increases beyond that intended by the designer of the equipment.

If feedback capacitors open or become larger, depending upon their location in the circuit, instability can arise because of excessive feedback or by additional phase shifts which the capacitor was designed to counteract.

If you have an AC VTVM, check the gain of each stage and find the one which is causing the trouble; then concentrate your search there. It may be helpful to disconnect the feedback circuit, lest it influence the gain. If all stages operate normally, you must then look into the feedback circuit. Measure the gain at various frequencies with and without feedback; if it is reduced when feedback is applied by more than 20 or 25 db, than excessive feedback is probably present. For an accurate appraisal of the feedback problem, consult the design notes of the equipment to see just how much feedback is supposed to be present.

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Crackling in Amplifiers

Q. I have on hand an old 50-watt PA amplifier. The tube lineup consists of two 6J7's, two 6N7's, four 6L6's, one 6X5, and two 5V4's. The nature of the trouble is a crackling noise in the output. I have replaced many resistors and electrolytics with no improvement. J. L. Cosette, Quebec, Canada.

A. First, check the tubes. This is always the first thing to do when servicing equipment, partly because they can cause a multiplicity of troubles, and partly because they are the easiest to check, especially when you have replacements. If tubes check normal, look to the coupling capacitors. Since your unit is an old one, it probably contains many waxed paper capacitors, and such units often give rise to the type of trouble you described.

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Hiss Level in Preamplifiers

Q. How can I reduce the background tube noise which occurs when the preamplifier is in the phonograph position? The noise shows up especially when solo instruments are playing. C. I. Schup, Lawndale, Calif.

A. Perhaps the background noise you notice is the result of running your power amplifier at excessively high level compared to the level of the preamplifier. In some instances, the stages following the preamplifier volume control have considerable noise content. If the gain of the power amplifier is set too high, too much of this noise will get into the amplifier. Simply reduce the gain of the power amplifier and increase that of the preamplifier.

Unfortunately some power amplifiers are not equipped with input gain controls. In such cases, you may find it advisable to modify your unit to include such a control. If, for some reason, you are unable to include the control, make up a voltage divider of fixed resistances.

Sometimes the noise results from poor signal-to-noise ratio in the phono stage of the preamplifier. The signal-to-noise ratio becomes worse as pickups with smaller and smaller outputs are used. If the trouble is in the phono stage, you can determine it by raising and lowering the volume of the preamplifier. If the hiss level changes as the control is rotated, the trouble lies in the phonograph stage. There is probably little you can do about this trouble except to use a cartridge whose output is higher than that of your present cartridge. Before doing this, however, check to see if the manufacturer of your present cartridge has a stepup transformer designed to work into preamplifiers requiring higher input drive.

If you are handy with a soldering iron, you can try replacing some of the resistances in the phono stage with others of larger wattage rating. This step may be especially helpful if the resistors now in the circuit are of half- or quarter-watt rating.

Distortion and Volume Controls

Q. My power amplifier is behaving in a most peculiar manner. When the volume control is turned up full, the bass response is normal, but as the setting of the control is reduced, the bass falls off sharply. This is not a result of the Fletcher-Munsen effect, for, as I advance the gain of preamplifier at the same time that I decrease that of the power amplifier, the bass still continues to fall off. Enclosed is a schematic of the input circuit of the power amplifier, (Fig. 1), *perhaps it will help you to determine what is wrong. James P. Gooley, Oak Lawn, R. I.*

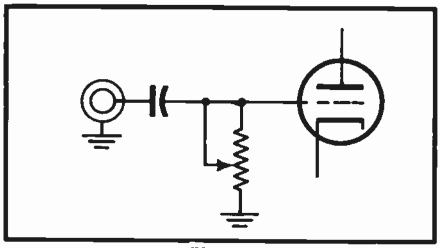

Fig. 1

A. Notice that in the input circuit of your amplifier the volume control is connected as a rheostat. This circuit can perform the function of reducing the gain of the amplifier, but, as you have seen, the bass will be attenuated in greater amount than the remainder of the signal as the gain is lowered.

To explain why this circuit causes this behavior, let us assume that the reactance of the coupling capacitor is 0.5 megohm at 30 cps, and that the resistance value of the potentiometer when fully open is also 0.5 megohm. At 30 cps, half the voltage developed by the preamplifier will be lost across the coupling capacitor, while the remaining half is available for application to the grid of the input tube of the power amplifier. Let us assume that the gain has been reduced, and the resistance of the control is now 0.25 megohm. Of course the over gain of the amplifier has been reduced 6 db, but our 30-cps tone has been decreased by an even greater amount. The reactance of the capacitor is still 0.5 megohm, but the resistance of the pot has been reduced so now only ⅓ the voltage produced by the preamplifier is available to the power amplifier. This effect will become more and more severe as the resistance of the potentiometer decreases. What is needed is a control circuit which maintains a constant reactance, but still allows the grid to pick off as much signal as is needed, by means of voltage divider action. *Figure*

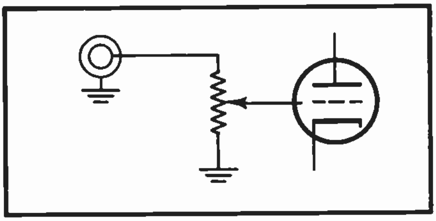

Fig. 2

2 shows how this same 0.5-meg. control can be wired to accomplish this. Note that the coupling capacitor is connected through the full resistance of the pot to ground. The slider has no effect upon the reactance presented to the coupling capacitor, since the grid to which it is attached draws no current and is therefore of infinite resistance. ÆE

Volume Controls and Power Amplifiers

Q. I have a power amplifier having an input grid resistor of 1 megohm. I also have two preamplifiers and I would like to connect a volume control to each of their outputs in order to improve signal-to-noise ratio. What value of volume control would you recommend that I use? Neither preamplifier has a cathode follower. Would it be better to locate the volume control at the end of the preamplifier or locate it at the input of the power amplifier? What are the general principles involved in determining the resistance of a volume control? S. W., New York City, N. Y.

A. First, remove the input grid resistor from the power amplifier. Connect a volume control of the same value in its place. (Later in this answer I will outline the means of determining the correct value of control. Check to see whether the control you substitute conforms to this procedure.) Be sure that the grid of the amplifier is connected to the arm of the control. The leads of the input connector on the power amplifier are connected across the full resistance of the control. The method of determining the value of a volume control is the same as that employed when determining the value of a grid-load resistor. It should be at least twice as large as its preceding plate-load resistor. The value of the interstage coupling capacitor is also a factor. Its reactance at the lowest audio frequency you wish to pass should be considerably lower than the grid resistance.

It would be better if the control units employed cathode-follower outputs. This is especially true when the control units are to be located some distance from the amplifier because the length of the interconnecting cable will then be great enough to introduce considerable capacitive reactance which, in turn, will cause some degradation of the high frequencies. This condition does not prevail with cathode-follower circuits largely because their impedance is much lower than the capacitive reactance of the shielded cable, and hence, the shunting effect of this reactance is negligible. ÆE

Amplifiers and FM Interference

Q. I live about three city blocks from a fairly strong FM station which does commercial broadcast music via multiplex. This FM signal has been coming through my music system for about two years. By juggling the various a.c. power leads, audio cables, and all concerned, I have been able to keep the amount of signal picked at a minimum.

I have just completed a stereo installation. The FM station is coming in much more strongly, and curiously enough, it is more pronounced in the "left" channel than in the other. In the right channel, I get an overdose of hum, which doesn't worry me. I have been fighting the hum war for years, and I am confident I can lick that. My question is, How can I prevent this annoying interference? Cameron Magnon, Tampa, Fla.

A. The first thing to ascertain is just where the signal is entering the equipment. To do this, first short the pickup leads. If the signal ceases, you have obviously located the source of entry, the pickup and/or associated leads. If the interference still persists, short the grid of the second stage of the preamplifier and so on down the line until you have found the point of entry of the signal.

You will then have to experiment with bypass capacitors. You probably can find one whose value is sufficiently large to shunt out the interfering signal and yet small enough so as not to limit the high-frequency response of the equipment.

If this method fails, place a choke in series with the offending lead. This choke can be made by winding 20 turns of No. 24 enameled wire around a 1-megohm 2-watt resistor. You may have to provide shielding for this filter arrangement. The filter can be rendered even more effective through the use of a second capacitor, arranged in the circuit in such manner that the circuit is a capacitor-input pi-type filter.

Should the interference still continue, you will have to resort to a wavetrap rather than to the filter system just described. This is made of a series-resonant circuit placed across the offending input circuit, and a parallel resonant circuit placed in series with the hot lead. *This unit must be shielded* and the shield returned to a good ground. Capacitors here should be variable, and their values are 10 μμf maximum to 1 μμf with their rotors open. Inductances are wound of No. 14 enameled wire on a ⅜-inch form. Each inductance should contain 3 or 4 turns. The inductance should not be closewound, but rather, should be spaced. After the coils are wound the forms are slipped out, leaving a self-supporting structure. After the wiring of the wavetrap has been completed, set the amplifier to the phono position or whichever position produces the interference, and adjust the tuned circuit for minimun signal.

If no such dip can be found, the coils in the network contain too much or too little inductance. You can determine which of these conditions prevails by compressing the turns. Compression increases inductance, expansion decreases it. If this procedure does not produce a null in the signal, you must then add or subtract turns.

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Power Supply Filtering

Q. The power supply circuit for an amplifier I plan to build calls for an 8-H. choke and an 8-μf capacitor. Because of space limitations, I desire to substitute a resistor for the choke, and increase the size of the capacitor. I would like to know the relative value of RC filtering as opposed to LC filtering. I have noticed that when a choke is used in filter circuits, the capacitors are usually of smaller values than those used with a resistance-type filter. Does the use of RC filtering have any adverse effect, such as increased 120-cps hum or poorer voltage regulation? R. D. Dickson, San Francisco, Cal.

A. The advantage of LC over RC filtering lies in the fact that smaller associated capacitors need be used because, in addition to resistance, the choke possesses inductance which tends to oppose any change in the strength of the current passing through it. The resistor in the RC filter merely serves as a time constant to slow down the rate of change and discharge of its associated capacitors. The choke has some resistance along with its inductance, and can therefore perform this time-constant function to some slight degree. If the capacitor and resistor sizes are made large enough, the ripple content of the power supply can be very nearly that of the LC type. Where small currents are involved, the RC filter circuit works very well indeed. Where the power supply is called upon to furnish current which consists of transient variations, such as in power amplifiers, the RC type filter is definitely *not* recommended, because the filter resistor is so large that, on peak current demand, much voltage is dropped across it. The voltage dropped across the resistor robs the load of the constant voltage needed for proper linear operation.

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Amplifier Power

Q. I've heard a great deal lately about amplifier power, to the effect that more is always better, provided that the power is clean at higher levels. Would this be true for a speaker system of high efficiency, such as a Klipschorn? I'm using a 30-watt amplifier to feed such a system but, if 50 watts of comparable quality would give me better performance, I would like to make the change. J. H. Moore, Tulsa, Oklahoma

A. So long as the power amplifier is of good quality and is operated well below its maximum capabilities and is fed into a speaker of reasonably good efficiency, there is no need to substitute one of higher power output capabilities for that which you are now using. However, if your listening environment is such that this amplifier must be run too close to its capacity, then I should certainly suggest that a more powerful unit be substituted. Be sure that the speaker system is capable of continuous operation at the highest program level to be used, and in fact, some tolerance should be left to account for transient peaks. If your speaker system cannot cope with the demands to be placed upon it, additional speakers should be used which can take up the power and which can provide better sound dispersion, too.

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Amplifier Instability and Remote Lines

Q. This past weekend I was asked to connect a remote speaker system for a friend.

The system consists of an amplifier feeding a 500-ohm line, thence to a 500-ohm speaker and an 8-ohm speaker fed through a matching transformer, each speaker controlled by "T" pads on the speaker side of the transformer. The distribution line was an unknown (but very long) length of No. 18 lamp cord.

At moderately loud volume levels, the system was unstable and would motorboat badly. Reduction of bass would stave off the motorboating some but not much. In connecting an outboard jack to the line I *noticed that holding one lead of the line while standing on wet ground encouraged the instability; touching the other lead had no effect. Also, this effect was not noticeable when standing on a dry board.*

The amplifier was connected to the speaker through a short lead and full gain could be used with no instability.

From the above I assume the instability is due to line capacitance. If so, could this be eliminated without replacing the line, since replacement would be almost impossible? H. S. Newins, Red Bluff, Calif.

A. I agree with you that the instability is caused by an alteration of the feedback characteristics of the amplifier resulting from the long line. It will be hard to say whether this trouble is the result of capacitive effects or inductive effects because a long line will contain significant amounts of both.

Before adding reactances and capacitances in an attempt to tune this difficulty, ground the amplifier to a good ground, and ground the common side of the far end of the 500-ohm line and one side of each speaker voice coil. Sometimes this kind of grounding will shunt out this kind of instability. If it works, it will save you much trial and error fiddling with inductances and capacitances which will otherwise be your fate.

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Connecting Headphones to Amplifiers

Q. I would like to know how to connect a pair of 600-ohm headphones to my Brook amplifier. John Sabritt, Philadelphia, Pa.

A. The Brook amplifier is a special case in that it includes a 500-ohm winding in its design. Furthermore, this amplifier is fitted with an octal plug, rather than the conventional tie strip used to connect loudspeakers and other devices to most other amplifiers. Pins 7 and 8 on this output plug represent the 500-ohm winding, and your phones may be connected directly to these pins.

However, there is another factor which should be taken into account. Earphones are placed very close to the eardrums, thereby creating a very efficient coupling network. This means that only a small amount of power is necessary for good listening level in the headphones. In fact, the power required for adequate listening level is approximately 9 milliwatts, or 9/1000 of one watt. This means that the volume control need be turned but slightly to produce such a low power level in its output circuit. This small degree of volume control rotation will make it difficult to operate your amplifier smoothly.

This difficulty may be overcome through the use of a pad. It may be constructed as follows: Connect a 500-ohm resistor across pins 7 and 8 of the output plug on your amplifier, the 500-ohm tap. Use a 5-watt wire wound resistor. Connect a 4700-ohm, 1-watt resistor to pin 7. Connect a similar resistor to pin 8. Connect the free ends of these resistors to the headphone terminals. By this means you will have introduced a loss into the system, a voltage division of 20 to 1. Naturally, considerably more power will be needed to drive the headphones with this pad in the circuit.

The pad has another advantage. Because of the high sensitivity of the headphones, any residual hiss level in the amplifier will be present as an annoying background to the program material being listened to. The pad will attenuate this background noise to a level such that it will be barely noticeable.

Other amplifiers are not provided with a 500-ohm winding, but these also may be connected to 600-ohm phones. The amplifier again should be properly loaded with a resistor connected across the output tap selected. If, for example, the 8-ohm tap is to be used, it should be of perhaps 5 watts capacity. A similar voltage-dividing network should be used between the 8-ohm resistor and the headphones. If you wish to experiment in order to obtain the best signal to noise ratio, make the resistances variable. Once you have selected the desired settings, measure the resistors and substitute fixed values.

The headphones may also be connected directly to the output of your preamplifier. 600-ohm headphones will unfortunately load down most cathode-follower circuits, and this loading will cause a degradation of low-frequency response, which is already badly degraded because of the nature of most of the headphones employed. High-impedance phones, especially crystals, may be connected directly to the output of the preamplifier without such degradation.

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Component Life

Q. *I have a Heath Kit W5M amplifier, and WAP2 preamplifier. When these components were new, the noise level was inaudible unless one searched for it. Now I am beginning to notice a little high-frequency noise. It has not come to an irritating level as yet. It is a case of where before there was none, now there is some. I am assuming that this noise is caused by the gradual breakdown of resistors or capacitors, as I have replaced all the tubes* *and the noise remains. How long should one expect carbon resistors, paper capacitors, electrolytic capacitors, ceramic capacitors, etc., to perform their functions satisfactorily? How long should one expect tubes such as those used in the equipment described, to last under normal conditions? Clyde A. McGoldrick, APO, San Francisco, Cal.*

A. The usable life of any component depends its initial quality and upon the conditions to which it is subjected. These might include temperature, moisture, and the degree of closeness to its maximum rated capacity at which it is operated. Resistors operated well within their ratings may last longer than ten years. Waxed paper capacitors may be expected to last about five years, although in many cases their lives are much shorter. With these components, temperature and moisture are most important. I should expect molded paper capacitors to last close to ten years when conservatively operated. Mica and ceramic capacitors may well last more than ten years. The life expectancy of electrolytic capacitors varies greatly. High-grade units may last five to ten years, and sometimes longer, while some units last barely a year.

Tubes are in a class by themselves, and hence will be given special treatment. When used for periods of perhaps an hour or two a day, they may be expected to last three to four years. However, they may last for much longer than that. I have had tubes in my FM tuner for ten years, and still others in my communcations receiver for 12 years, and they are still performing satisfactorily. Rectifier tubes are usually the first to burn out.

Tube life may be greatly lengthened if the proper measures are taken before the tubes are placed in service: Place the tube in a convenient and appropriate socket. Many of you can salvage it from an old chassis or from the junk box. Preferably, the tube should be in a vertical position. Connect the heater terminals to a source of power whose voltage can be varied. Over a period of a week gradually advance the heater voltage from zero to normal. Do not apply plate voltage. Allow the heaters to remain at their normal voltage for another week. The longer the tube has stood on the shelf, the greater will be the need for this aging process. Tube manufacturers have neither time nor facilities for this kind of aging. They connect the heaters of the tube, for a short time, to a source of voltage higher than that recommended for normal use. This process, because of its extremely brief duration, is known as flash aging. Its purpose is to cause the movement of more electrons to the surface of the cathode, thereby improving the emission capabilities of the tube.

Another aid to tube life is never to turn off the equipment. Remember, though, that the money saved on tube replacements will be much less than the electric bill run up by leaving the equipment running.

The data presented here represent information gained while servicing equipment.

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Tuner Problems

Q. 1. The AM section of my tuner produces more hum and background noise than the FM. I have heard other tuners in which the AM section is quiet. 2. When I connect the FM section of the tuner to my double conical TV antenna, I receive the same station at many places on the dial. What is causing these two conditions and what can I do about it? Robert McDonald, Oakland, Calif.

A. 1. The hum which is present in the AM section of your tuner can be caused by several things: perhaps the AM section is not well filtered, leading to the supposition that perhaps one of the filter capacitors has become defective. There may be a leak between the heater and cathode of one of the tubes. It may be generated as a result of poor grounding or oxidized house wiring. If this latter is the case, I don't believe there is much you can do, especially if you are an apartment dweller as I am. There is also the possibility that the hum is caused by something on the line to which the tuner is connected. I have an AC-DC dictating machine which, when turned on, introduces hum into every AM receiver in the house. Hum arising from these last two sources can sometimes be minimized by the use of an outside antenna. It need not be elaborate. Make it about 20 feet long and keep it well insulated from surrounding objects. The lead-in wire from the antenna should be of coaxial cable, so that the lead-in cannot pick up any interference. Naturally, the shield of the cable should be returned to a good ground, as should your tuner chassis.

The background noise of which you spoke may have its roots from many places. Manmade interference, such as that produced by vacuum cleaners and fluorescent lights, probably heads the list of possible candidates. (Next month I shall discuss a special type of background interference, that of the direct radiation of harmonics of the horizontal oscillator.) Background noise can be generated within the tuner itself as a result of defective tubes, coils, r.f. and i.f. transformers, resistors, capacitors, and the like. The stage which, in my experience, most often causes this trouble is the mixer stage.

2. The reason you receive one station at several places on the dial is that the front end of your tuner is misaligned, poorly designed, or overloaded by excessively high input signal levels. I suspect the latter possibility because your antenna is very efficient. A straight dipole or folded dipole will probably give more than adequate results. Keep the dipole out-of-doors if possible because, when it is indoors, passersby may cause the desired signal to be reflected away from it, leading to fading and fluttering.

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Delayed AVC

Q. What is delayed AVC and what is its purpose? George Lystad, Lake City, Ark.

A. The purpose of AVC is to keep the audio output of the receiver relatively constant, regardless of the strength of the received signal. This is accomplished by rectifying the signal voltage and connecting it in such a manner as to make the grids of the r.f. and i.f. stages more and more negative as signal strength increases. Thus, the receiver is more sensitive to weak signals than it is to strong ones. No matter how weak the signal, some voltage is fed back, thereby reducing the sensitivity of the receiver. Obviously, when reception of weak signals is to be accomplished, the receiver must operate at maximum sensitivity. This cannot be done when the AVC is operating, as previously noted. Therefore, means must be provided to make the AVC operate only when signals reach at least a moderate strength. This can be done with a manual switch connected in the AVC bus which removes it from the tube circuits. There is also an automatic means for accomplishing this. Two diodes are needed for this circuit. One is connected in the conventional manner, and is used to demodulate the signal. The other is used only to develop AVC. The plate of this diode is biased negative with respect to its cathode by the desired amount. Weak signals will not develop sufficient voltage to overcome this bias but, as signal strength increases, the diode will conduct and form AVC in the usual manner. Since the AVC is not operative until the signal reaches a predetermined strength, this system is known as delayed AVC (DAVC). The use of two diodes is mandatory here, since if only one were used, it would fail to conduct at all on weak signals, and even when it did start to conduct, less than half of the cycle would be reproduced, leading to distortion.

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Dynamics, AVC, and Broadcasting

Q. I believe I understand the principle of the action of AVC circuits in radio receivers. However, one thing puzzles me. Wouldn't this action operate on the signal itself and compress the dynamic range of the program material. I have never yet heard a radio broadcast that sounds like a phonograph; there is always some sort of degradation which transmitter and receiver distortion does not always explain. William Devine, Detroit, Mich.

A. When AVC is used with AM receivers, the filter time constants are so chosen that the rapid modulation peaks are not smoothed out, but slow variations in carrier strength are. When the dynamics increase, the average strength of the carrier is maintained, whereas the peaks and troughs increase in size. If the time constants are chosen to be 100 milliseconds, frequencies as low as 20 cps can be transmitted without the AVC action smoothing them out unduly. As the time constant is shortened, more and more of the lows will be smoothed out, leading not to a reduction of the dynamic range, but to an erasure of the lows from the audio output.

When the volume varies during FM broadcasting, the sizes of the peaks and troughs remain constant. Volume increases manifest themselves as increases in frequency deviation from the center frequency. It is these frequency deviations which are detected, rather than any change in carrier amplitude. It should be obvious now that the clamping effect of the AVC and limiters can in no way impair the dynamic range of the program material.

Compression of dynamics probably comes from the broadcaster's desire to maintain as high a signal-to-noise ratio as possible.

Tuner Sensitivity and Quieting

Q. *Can you tell me whether it is possible to have an FM receiver which will completely suppress ignition noise interference even when the latter appreciably exceeds the desired signal? If not, what type of circuit most nearly approaches this ideal? Can you explain exactly how the sensitivity in mv for 20 db quieting is obtained for FM tuners? Judging by the advertisements in* AUDIO *and other magazines, there are two or more methods in use. What does absolute sensitivity mean? What is the theoretical limit of sensitivity of an FM tuner? B. H. Murdoch, Belfast, Northern Ireland*

A. The only answer I can give with regard to a tuner having perfect suppression, is that there is no such thing as perfection. Any limiter can be upset when overloaded with noise signal. The amount of signal needed to accomplish this will depend upon both the strength of the desired signal and upon the design of the limiting circuit. Probably the closest approach to this ideal is the discriminator circuit, preceded by two limiters. Following very closely behind this is the ratio detector preceded by a single limiter and containing stiff AGC. As a matter of fact, the discriminator circuit should also employ some AGC, especially at the front end. This is because a tuner must handle signals of greatly varying strength. In the New York area, for example, it is possible to receive signals as strong as 0.5 volt (yes, ½ volt) at the antenna terminals. At the other extreme, the tuner must accept signals as small as 2 microvolts and still manage to quiet satisfactorily. That's asking a lot of a front end, and it is the reason that AGC is very much needed. Notice that with the discriminator circuit, two limiters are needed, whereas with the ratio detector, only one is needed. This comes about because the ratio detector circuit has inherent limiting properties. Further limiting would only reduce i.f. gain, and this would serve no useful purpose.

Now, let's go into the problem of sensitivity and quieting. There are several methods for measuring these two quantities. When these are applied properly and interpreted correctly, they mean much the same thing. The standard employed by the Institute of Radio Engineers may be summarized as follows: What voltage, when fed into a 300-ohm input, will give 30 db quieting, when 22.5 kc deviation is applied? Notice that there are two other terms which must be taken into account besides input voltage and the number of db of quieting. These are the input impedance and the amount of deviation, or percentage of modulation. If we make our measurements at an impedance of 72 ohms and feed the same power to the input, the voltage appearing at the antenna terminals will be half that which would be obtained at the 300-ohm impedance. This means that if our tuner requires four microvolts for 30 db of quieting at 22.5 kc deviation and at an impedance of 300 ohms it will need only two microvolts for the same degree of quieting when an impedance of 72 ohms is employed. This sounds like an improvement, but a 72-ohm antenna system gives us only half the signal voltage provided by a 300-ohm antenna system. Naturally it is assumed that both antennas are of equal efficiency; and that they are in identical locations. In other words, the two measurements are, for all purposes, identical.

Next, we come to the matter of deviation. The I. R. E. used 22.5 kc because it corresponds to 30 per cent modulation. This, in turn, is roughly equal to average program level whose peaks are 10 db higher, equaling 100 per cent modulation. This figure was selected because people listen to average program level most of the time, rather than to peak levels. The Institute reasoned, therefore, that noise impairs average level more than it does peak levels because the average signal strength is weaker than peaks. Other methods, however, make use of a deviation of 100 per cent, 75 kc, as the basis for their quieting measurements. This gives us an apparent improvement of slightly more than 3:1. Our tuner which required four microvolts for 30 db of quieting will now require only 1.333 microvolts and actually slightly less, for 30 db of quieting. If we use an antenna system and input circuit designed for 72 ohms impedance, we will have a tuner requiring only 0.666 microvolt for 30 db of quieting. That sounds like a pretty good tuner, but it's no better than our original model, or should I say, "no better than our original specifications," since we have really done nothing at all to the tuner. Actually, it's all in how you interpret the figures.

We can go even further in our direction of smaller and smaller input voltages for good quieting. All we need do is to assume that good limiting can be had with 20 db suppression, rather than 30 db. We need only ⅓ as much signal to obtain this degree of suppression, and our figures are growing small indeed, but then, so is our suppression. There are other factors which affect suppression.

There are other factors, however, which caused the standards committee of the Institute of High Fidelity Manufacturers, Inc., to propose a new set of standards. They note that as the signal strength decreases, not only does the noise increase but so does the distortion. This increased distortion is largely caused by a narrowing of the i.f. band-pass. Therefore, the standards committee of the Institute of High Fidelity Manufacturers conceived the idea of a total usable signal measurement. The method for making this measurement can be summed up as follows: What signal voltage, at an impedance of 300 ohms, and at a 75 kc deviation, will be required to cause a signal at the output of the tuner which shall consist of 3 per cent total noise and distortion? The method described takes both these factors into account, and this method is quite valid. However, it certainly is going to add much confusion to already troubled waters.

Lastly, you wanted to know what is meant by absolute sensitivity. It is approximately 0.71 microvolt for 20 db of quieting, at a deviation of 22.5 kc. The reason that no greater sensitivity is possible is that the input circuit will contain noise of its own which will be 20 db below this value. It should be stated that this measurement is based upon an input impedance of 300 ohms. Any impedance generates its characteristic amount of noise, and there is no way we can prevent this, unless this impedance were placed at a temperature of absolute zero, which would of course cure the random noise generated across the impedance, but so placing the impedance would pose grave problems for the tuner manufacturer.

When making any measurement where impedance is a factor, it is important that the input to the tuner be truly matched if valid results are to be obtained. Most signal generators have outputs of 50 or 72 ohms. Equal values of resistance should be placed in each leg of the generator, and the total should equal the impedance of the tuner input. Only balanced generators should be used, since an unbalanced unit will introduce standing waves which will affect the validity of the measurement. If an unbalanced generator were used, however, all the padding resistance would have to be placed in the hot side of the line.

Interference from TV receivers

Q. *When I tune my receiver through the AM band, I find it covered with a series of whistles whose pitch varies according to the station to which I am listening. Since this phenomenon occurs with many other receivers in the apartment house, I don't think it is because of any possible misadjustment of my whistle filter. The only time the whistles do not appear is early in the morning. When my TV set or someone else's, is turned on, the whistle reappears. What is causing this, and what can I do about it? Woodrow C. Doebler, Pottstown, Pa.*

A. This series of whistles is the result of beats between broadcast stations and harmonics of the horizontal oscillator of the offending TV receiver. Since the space between harmonics is small (the frequency of the horizontal oscillator being 15,750 cps), they can beat with any station.

There are two ways by which the harmonics can enter your receiver. The first of these is by direct radiation from the TV set, and the harmonics of the oscillator are picked directly by the receiver's antenna. This effect can sometimes be minimized by placing your antenna high enough so that it will be out of the radiation field. (Some receivers have been known to radiate several hundred feet, in which case you cannot erect an antenna high enough.) This can also result in the strength of the desired broadcast stations' being great enough to override the whistles. The antenna should be fed by coaxial cable, so that no TV set radiation will be picked up by the lead-in.

The second and less unlikely means of entry of these undesired signals is through the power lines. This may be overcome by inserting an r.f. choke in each side of the line feeding your tuner, and bypassing all choke leads to ground with good-quality mica capacitors.

Radio Volume Control Considerations

Q. *When the volume setting of my radio receiver is changed rapidly, I notice a considerable delay between the time I make the volume change and the time when the change becomes audible. What could cause this? Madelain Gold, Chicago, Ill.*

A. *Figure* 1 shows a typical volume control circuit of an AM radio receiver. The potentiometer, R_1, is a portion of the diode load and is therefore carrying a steady

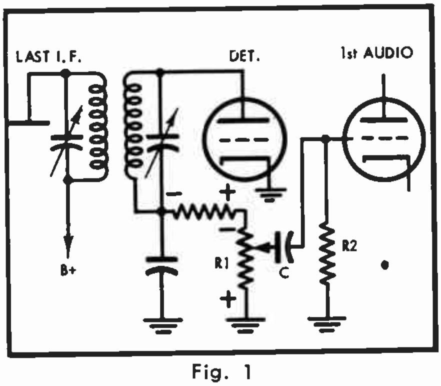

Fig. 1

direct current. As the control is advanced, a greater and greater charge is applied to coupling capacitor *C*, which charge is transferred to the grid of the first audio stage. If the grid resistor for this stage, R_2, were infinite, this charge could never leak off, and the tube would be cut off. As R_2 is made smaller, the charge can leak off more and more easily. The size of the coupling capacitor also determines the time needed to discharge through R_2. If the capacitor is large it will have to take more time to discharge, since it took a greater charge initially. If the discharge rate is low there will be a negative voltage applied to the first audio grid every time the volume control is turned up, which may completely cut off the stage until balance is once again reached. Some may say that the capacitor is intended to block d.c. from ever getting onto that, which is true. Remember, however, that, so long as the capacitor is being charged, it does not exist. The process is something like having a resistor substituted for the capacitor. When the d.c. is first applied, the resistance is zero, gradually rising to infinity, as the charge builds up to its maximum. In practice infinity is never reached, since there is always a certain amount of leakage present within the capacitor itself.

Matrixing

Q. *I have heard much lately about matrixing. Just what is it? G. Best, Long Island City, N. Y.*

A. Matrixing is a system whereby two signals are combined to form their sum and their difference; later they are reconstituted into their original components. This technique is employed in the Crosby multiplex system. The sum of two stereo channels is fed into an FM transmitter in the normal manner. (Those who don't have multiplex adapters may, by this means, receive the monophonic broadcast.) The difference signal is impressed upon a subcarrier, which is transmitted on the main FM carrier. This subcarrier is undetectable with unmodified FM receivers, but this difference information can be recovered by adding a multiplex adapter to the tuner. Some circuit modification will be needed on tuners which were not provided with a multiplex jack. One of the things which is included in the adapter is a matrixing circuit, which serves to recover the difference signal and recombine it with the sum signal in such a way that the original stereo information is recovered and can be fed to the stereo preamplifier in the usual manner.

Another use found for this technique is the Columbia Record stereo system, which matrixes the signal and records the sum signal laterally and the difference signal vertically, with compression on the difference, or vertical channel.

Multiplexing

Q. *I recently bought an FM tuner on which there is an output marked "multiplex." Can you tell me to what purpose this may be put? R. P. Burns, Havertown, Pennsylvania.*

A. Multiplexing is a system by which two or more programs may be transmitted simultaneously by the same FM transmitter. In one of the several methods now in use, one of the two channels is transmitted in the normal manner, while the intelligence of the other channel occupies a bandwidth of 15 kc. in the supersonic range. This channel is separated from the normal channel at the detector by means of a sharp cutoff high-pass filter. After this separation process, it is heterodyned with another oscillator in such a way that the beats fall into their pitch relationships in the audio spectrum, and can therefore be heard. This is just one of several methods which are in use, or which are under study, for the simultaneous transmission of two or more programs. Until one method is adopted as the standard to be used by all interested parties, little, if any, equipment will be available for detecting the second channel.

The multiplex output is wired to the detector circuit in such manner that the de-emphasis network is bypassed. If your amplifier is poor in high-frequency response, you may connect it to the multiplex output. FM reception will be normal except for an exaggerated emphasis of the highs, which can be helpful in systems where response in that region of the spectrum is poor. Let me make it perfectly clear that connecting your tuner in this manner will not allow you to hear the multiplexed channel. You will hear the FM programs to which you are accustomed, but with an overabundance of highs.

Remote Cartridge Circuit

Q. My layout is such that I keep by Rondine hysteresis motor and Pickering cartridge in another room, quite removed from the amplifier. As the leads are fairly long, I fear considerable loss of highs in the shielded cable. Could you suggest a simple circuit using, perhaps, a 12AY7, with one stage of flat amplification and a cathode follower stage using d.c. on the heaters? Frank Gittelson, Lynbrook, N. Y.

A. *Figure* 1 should solve your problem.

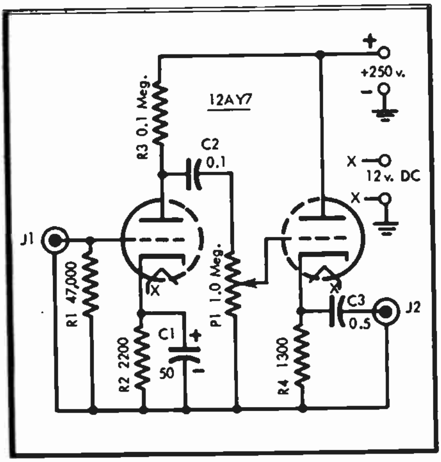

Figure 1

The circuit employs a 12AY7, chosen because of its low noise content. It is used as a voltage amplifier and cathode follower output. Thus, the pickup is presented with its proper impedance and the output impedance is sufficiently low to enable the use of lines up to 50 ft. in length or even more, without serious high frequency degradation. Because of the possibility of overloading your preamplifier, a volume control was incorporated in the cathode follower stage to limit the signal feeding the preamplifier to a level somewhere below the fold-up point. The circuit is designed to operate with approximately 250 volts on the plates. Since the plate and filament supplies for this circuit are entirely conventional, they are not shown. This circuit may also be used to feed other high-impedance devices into the preamplifier. Care must be taken not to overload the 12AY7 and means must be provided for adjusting the value of the input resistor, should the output impedance of the device used be in excess of 47,000 ohms. Æ

Matching Transformer Impedances

Q. The impedance of my phonograph cartridge is very low, resulting in output voltage insufficient to drive my preamplifier. I therefore resorted to the use of a matching transformer. However, I noticed that when the secondary is terminated in a resistance of the proper value, less output is obtained than is had when the transformer is terminated in a resistance several times larger than that of its nominal secondary impedance. What is the significance of these findings? S. Kalmer, New York City

A. We match transformer impedances mainly with the idea of transferring as much power from one circuit to another as possible. This can be done only when the transformer is terminated in a resistance equal to the nominal impedance of the transformer. However, a vacuum tube is not a power sensitive device but is, rather, voltage actuated. A transformer will, when terminated in a resistance higher than its proper terminating resistor deliver a higher voltage into that resistance than it would into its correct value. Although this happens, maximum power is no longer transferred. While maximum voltage will occur when the transformer is terminated in a resistance at least five times its nominal rating, the low-frequency response is optimum only when the transformer is properly terminated, since the low frequency response is determined in part by the current flowing in the windings of the transformer. Exactly what will happen to the response varies from unit to unit and so, no figure can be given. If the additional gain is not required, terminate the transformer in its proper load. If it is needed, adjustment of the bass controls should be carried out by means of trial and error.

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Stereophonic or Monophonic Sound

Q. I have many monophonic discs and tapes. Now, with the advent of stereo, is my monophonic collection worthless? Arthur Darrow, Albany, N. Y.

A. I have met many people who lament the fact that stereo has made their fine record collections obsolete and miserable. While it is unquestionably true that the addition of spaciousness to music adds much to our emotional reactions to it, it does not and should not mean that we can no longer enjoy our otherwise fine discs. When you stop and think about it many people began collecting records in the 20's and before, and the sound on those early discs was poor indeed compared to those of today. Those people have not discarded them. I guess this is partly related to sentiment and is the symbol of a past which many of them considered to be better than the present. Probably, though, in the vast majority of instances, people hold onto these discs simply because of the artistry of those appearing on them. Stereo, wonderful though it is, cannot give us those oldtimers who have flashed across the concert and popular stages. This is not simply true of the 20's. It holds even for comparatively recent monophonic releases. What about those releases of old 78's or the immortal performances of Toscanini? True, stereo could have enhanced all of these performances, but they are still fine, valid ones, even without stereo. Of course, a record need not be world-acclaimed in order for it to be enjoyable to you. The main thing is that you liked it when you bought it and probably did right up until the time you heard your first stereo broadcast. Listen to that monophonic disc or tape again and you will probably enjoy it as much as always. I have such discs and tapes in my own collection, and I have stereo as well.

If possible, you should have equipment capable of playing both types of material. After all, there will be new music and new performances of old music. All of this will be recorded for our enjoyment. It will be captured in stereophonic sound.

You should not feel that the stereo system is just a flash in the pan, since material is coming at us with extreme rapidity, and this material and the equipment with which to play it, is being sold at a tremendous rate. It is hard to believe, but in the short time that the stereo disc and tapes have been with us, over 1,000 titles have been released.

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Distortion in Monophonic Discs

Q. I have been a record collector for a good many years, and, as you can imagine, have a great number of monophonic discs, in addition to a few of the new stereophonic discs. Because this stereo system is compatible, I use my stereo cartridge to play both types of discs. However, many of the monophonic discs are heard with a surprising amount of distortion as contrasted to their sound when a conventional, monophonic cartridge is used. What is causing this distortion and what can I do about it? James Blake, San Francisco, Cal.

A. There are several possible causes for the distortion of which you speak. One is that you are using insufficient tracking force. Some stereo cartridges are constructed in such a manner that they must track at or slightly above the prescribed force in order for the stylus to be properly oriented with respect to magnetic pole pieces. Failure to observe this precaution means that the cartridge will be too compliant, and the stylus will ride out of the grooves, rather than tracing them faithfully.

The second cause of distortion is a function of the monophonic discs themselves. Many of them are greatly overcut, especially at the high end of the audio spectrum. This will result in the inability of the stylus to stay in the groove properly. It tends to ride up out of the groove, regardless of tracking force. Any time that the stylus rides upward some vertical signal will be produced. Because of the nature of the system, this vertical component will transmit it to the loudspeaker. Most monophonic cartridges were specially designed to eliminate as much vertical output as possible, and for this reason, little, if any, of this form of distortion was detectable with your monophonic cartridge.

This trouble can be minimized by connecting the two sections of your cartridge in parallel, phased in such a way that the vertical output is cancelled. Most instruction sheets supplied with these stereo cartridges show a wiring configuration which will bring about this end. Naturally, if you wire this cartridge in this manner, you cannot achieve the stereophonic effect. What you will have to do is to wire a switch and mount it in some convenient place. This will enable you to switch from stereophonic sound to monophonic sound. If there is sufficient interest, such a switching circuit will appear in a future column.

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Turntable Speed

Q. I have heard much discussion about how the size of the intermediate idler of a phonograph turntable will affect the turntable's speed. Is this so? I have seen tables which slow down as the idler is pressed more firmly against the motor shaft, thereby making the idler smaller. Mark W. Tennberg, Albany, New York

A. The size of the idler has no effect upon turntable speed. The reason a table slows down when more pressure is created between the idler and the motor shaft is not that of the idler's possible reduction in size, but simply because the motor is loaded more heavily under those conditions, with this loading, in turn, causing the motor shaft to turn more slowly.

In order that I may give reasons for my position, picture a motor shaft whose diameter is one inch. This, in turn, drives an idler whose diameter is one inch, which, in turn, drives a turntable whose rim at the area of contact with the idler is also one inch. Because the ratios of the motor shaft, idler, and turntable are 1: 1: 1, all wheels will rotate at the same speed. Assume now, that the idler has worn down to a half inch in diameter. The ratios are now 1: ½: 1. This means that, for every revolution of the motor shaft, the idler to which it is friction coupled must revolve twice. It would seem that the turntable must rotate faster under this condition. This is not true. The turntable is twice the diameter of the idler, which means that the idler must revolve twice in order that the table may complete one revolution. In order for the table to make one revolution the motor shaft makes one revolution, the idler now makes two revolutions, while the table makes one.

Should it still be hard to picture the foregoing, think of the parts in terms of their linear distances, which are their circumferences. Assuming no slippage, the number of linear inches traveled by a given point on the motor shaft must equal the number of linear inches traveled by points on each other part of the chain. Work out the ratios in terms of linear inches which must be covered by each part, and you will see that the turntable has covered the same number of inches, regardless of idler speed.

Do not, as so many people try to do, apply this explanation to belt-drive systems in which the motor drives a pulley which is belt-driven to another pulley, on the same shaft with which is located a third pulley belt-driven to a fourth, and so on. Each belt must be considered a closed, separate loop. In order for the above discussion to be applied to belt drives, it would be necessary for the intermediate pulleys to be common to two belts. This also applies to stepped idlers—those that have two or more different diameters.

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Ghost and Echo

Q. What are ghost *and* echo*? Jay Sharpe, Oakland, Cal.*

A. These are annoying disturbances which can occur as over-recorded discs are played back. When an instantaneous lacquer disc is cut, a spiral is cut into the surface. As signal is fed into the cutting head, the cutting needle moves laterally in accordance with the frequency and amplitude of the program material. Because of this motion, the spiral generated is no longer uniform, so that at certain times the grooves are closer to each other that at other times. Lacquer is fragile, and so, if the sideward motion is made too great (over-recording), the wall of the preceding groove is broken through, or at least deformed slightly, and the playback stylus will not track properly. If the wall is not actually cut into, it can become distorted, with this distortion taking the form of the modulation envelope being impressed. In this manner, the groove immediately preceding the one in which over-recording occurred will have both the impression of the signal originally intended and that of the signal intended for the groove following. This latter impression is not as loud as the desired signal, but is nevertheless quite audible. These faint tracings are known as ghost.

Echoes are created as the cutting needle swings in the direction of what will be the succeeding groove. The needle hits the land on this half-cycle so violently that internal pressure is built up, whose amount varies in accordance with the modulation of the groove being recorded. Most of the time, this pressure is removed when the following groove is cut, but in some instances it remains and distorts the newly cut groove wall to the shape of its predecessor. Playing back such a groove reveals a faint trace of the material just heard. This faint after-sound is known as *echo;* it is necessarily far more rare than ghost, because of the tremendous force needed to distort uncut acetate or lacquer.

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Tape Recorder Boss Response

Q. I have made many tests with tape recorders, and I have noticed that none of those I tried have much bass below 30–40 cps. Is this because the tape is unable to accept frequencies below these values? James C. Bryson, Redwood City, California

A. The tape can accept frequencies as low as necessary, provided we take proper care in designing our associated recording and reproducing amplifiers. Characteristically, tape rolls off at the rate of 6 db per octave of frequency decrease. To correct for this we must design our reproducing equipment so that it possesses a boost of 6 db per octave of frequency decrease. This will balance the rolloff and we will have a flat response. Most amplifiers, however, are designed to boost 6 db per octave but only down to a certain point, say 50 cps, after which point the amplifier behaves as though it were flat, providing no further boost. This means that from there on, the bass rolls off at 6 db per octave.

There is no reason why the tape cannot accept tones lower than 30 cps. To illustrate this, let us assume a tape recorder running at a speed of 30 ips. We feed a test signal into the recorder at 100 cps. When the tape is played back at the same speed, we obviously have a tape whose program content is 100 cps. If the tape were slowed down to 15 ips, a 50-cps tone would be noted. If again the tape is played back at 7.5 ips, a 25-cps signal will be picked up, but you'll have to turn up your gain to get it, because your playback amplifier is no longer boosting complementarily to the tape's rolloff characteristic. 3.75 ips will give us a 12.5 cps tone, while a tape speed of 1.875 ips yields a tone whose frequency is 6.25 cps. This speed reduction versus frequency could be carried further till a speed of 0.0000001 *inch per minute* is reached, but I think it should be clear that we can, if we want to, arrive at any low-frequency limit we wish, so long as we design an amplifier with sufficient correction.

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Tape Direction

Q. *In* AUDIOCLINIC *in September, 1957, discussing conversion to stereo, you say that with the shiny side of the tape facing the viewer, the upper half is intended for the left speaker. Which way is the tape running: from left to right or from right to left? Paul M. Gerhard, Beverly, Mass.*

A. The upper channel always feeds the left speaker as the listener faces the speakers in listening position, and it is assumed that the tape is travelling from left to right.

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Microphone Phasing

Q. *I own a microphone mixer capable of handling the outputs of two microphones. The mixer is quite stable in that the channels do not interact with each other. I determined this by connecting a microphone to one channel and, while leaving the other input open, rotating the gain control to maximum while talking into the microphone. Rotating the control caused no change in the output of the driven channel. I checked the other channel with the same result. Yet, when I connect microphones into both channels and talk into both of them, increasing the gain of either channel causes the output to decrease. What would cause this undesirable effect? Raymond E. Leonard, Poughkeepsie, N. Y.*

A. In all probability, the microphones are out of phase with each other. If this is so, output from both channels will combine in the output of the mixer in such a manner as to cancel each other partially. This condition is easily remedied by reversing the leads to *one* of the microphones. In the case of microphones which make use of two-conductor shielded cable, interchange the connections of the conductors, leaving the shield connected to its grounding point. If the microphones have selectable directional characteristics and are close enough to be in the same sound field, they should be set to have the same pickup pattern. When bidirectional microphones are employed, be sure that all action takes place on the same side of both microphones.

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Spurious Response in Recorders

Q. *When a high frequency sine wave is recorded and played back through the recorder, a spurious response is heard which increases in strength in proportion to the fundamental, when the recording signal strength is increased. The frequency at which this phenomenon begins depends upon the quality of the recorder, and I have heard it start at 10,000 to 15,000 cps. I'm rather doubtful that this is a heterodyne effect occurring between the bias oscillator and the input signal, since I heard a response of estimated 5000 cps in addition to the 15,000-cps sine wave tone fed into the machine's input, when recorded on an Ampex 601, which has a bias frequency of 100 kc. If this were a heterodyning effect it would require approximately a sixth harmonic of the 15,000-cps input signal to combine with the bias frequency to give even a 10,000-cps spurious response, which seems unlikely. What is your explanation of this phenomenon? Burton W. Byler, Oreland, Pa.*

A. The 5000-cps tone you hear when a 15,000-cps signal is fed into the tape recorder is caused by the combining of a harmonic of this tone with the bias frequency. Despite the fact that the oscillation fed into the machine is a pure sine wave, harmonics may nevertheless be generated. The greater the level of the recording signal, the greater the harmonic distortion generated within the recording amplifier, and at the head where the signal is mixed with the bias. The bias frequency, furthermore, is only approximate. It is not at all impossible for it to be either 95 or 110 kc rather than the specified 100 kc. These frequencies can heterodyne with either the sixth or seventh harmonic, with the resultant 5 kc tone you have observed. It is also possible for the audio oscillator to be inexact as to frequency. A little figuring will show that an error of slightly less than a kilocycle can produce sufficient error at the sixth harmonic to cause the 5-kc tone, assuming that the bias frequency is exact.

Should you wish to determine precisely the frequency of the bias oscillator, couple some of its signal into the antenna terminals of a standard broadcast receiver, being careful not to introduce d.c. into the antenna coil. You will observe that a beat will occur at several places on the dial. This beat is caused by a harmonic of the oscillator combining with one of the stations. Where there is no station to beat against, the bias oscillator signal will appear as an unmodulated carrier. The frequency separating the appearances of successive signals from the oscillator is determined by the frequency separation of two successive harmonics, which, in turn, is equal to the fundamental frequency of the bias oscillator. The more accurately calibrated your receiver dial is, the more accurately the frequency of the bias oscillator can be determined. Of course, when two successive harmonics of the bias signal beat with two broadcast stations each of whose frequency is known and when each of the beat frequencies resulting from said combination is known, the exact bias frequency can be obtained without accurate receiver dial readings.

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Tape Hiss

Q. *What is the exact cause of tape hiss? This objectionable noise is present on many recorded tapes and even on some professionally-recorded original tapes. Burton W. Byler, Oreland, Pa.*

A. There are two basic causes: 1) The recording amplifier may contain tube and resistor noise whose volume is sufficient to cause it to be recorded on the tape in the form of hiss. 2) Magnetic tape may be considered to be composed of an almost infinite number of minute magnets. The number of these whose polarity and strength is used determines the nature of the recorded material. Some of the magnets are not used; their poles are aligned in a helter skelter manner. Because this alignment is non-uniform, it is obvious that small random groups of molecules will be aligned in the same direction, thereby combining to form larger magnets whose strengths are sufficient to cause voltages to be developed as the tape is being reproduced. These voltages are heard as hiss.

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Crossover Distortion

Q. *What is crossover distortion? William Adsen, Tampa, Fla.*

A. Crossover distortion is not, as might be supposed, an alteration of sound created within a network used to divide the frequency spectrum for use with two or three-way speaker systems. To make clear what crossover distortion is, we must reexamine some of the basic ideas concerning class-A and class-B amplifiers.

The tubes in a class-A amplifier operate approximately midway between the point where grid current flows and cutoff, where plate current flow ceases. This is a static condition which changes when a signal is applied to the grid circuit of the stage. At this time, the current in each tube no longer is equal to the current in the other tube of the push-pull pair. During the first half cycle, the current in tube 1 increases, while that of tube 2 decreases. During the opposite half cycle, the roles of the tubes reverse. The signal magnitude is such that the tubes are never driven into grid current, nor run down to plate current cutoff.

The class-B amplifier poses an entirely different problem, since the tubes are biased to cutoff. When a signal is applied to the grids of this stage, the following happens. During the first half cycle of signal, the grid of tube 1 becomes more and more positive, allowing more and more plate current to flow. As the signal voltage rises still higher, the grid becomes positive with respect to its cathode, and therefore draws current from the electron stream. The grid of tube 2, on the other hand, is driven more and more negative with respect to its cathode. The grid of this tube is already biased so far negative that no plate current can flow, and so this additional amount of bias causes no change in the operation of tube 2. As the polarity of the signal reverses, the roles of the tubes also reverse. It is obvious that in the class-B amplifier, only one tube at a time is operating.

If the tubes are biased to a point even more negative than cutoff, even by a slight amount there will be a point (where the signal is transferred from one tube to the other) where neither tube is handling the signal. This clearly is a form of distortion. Even when the tube is finally conducting, some distortion is present because a rise in grid voltage does not produce a rise (corresponding) in plate current near the region of cutoff.

The point where the signal is transferred from one tube to the other is known as the crossover point, and therefore, the distortion produced at this time is known as crossover distortion.

Nearly all high-fidelity vacuum-tube amplifiers operate at a point somewhere between class A and class B, usually closer to A. This condition is known as class AB_1. Because of this, the topic just discussed would have little more than academic interest to us, were it not for the introduction to the audio field of transistorized power amplifiers, which may contain one or more class B stages. They are used because they are more efficient than class-A circuits, since, when no signal is applied, no appreciable current flows. This greater efficiency leads to cooler operation, which is necessary to prevent excessive heat from damaging transistors.

Crossover distortion is minimized in these circuits by large amounts of feedback which make the base-collector relationship more linear.

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Crossover Networks

Q. I have seen very little published information concerning crossover networks. Would you supply information so that I can tackle construction of them? What is the reason for the use of such networks? Ray E. Roehrick, East Chicago, Indiana.

A. Research has not yet disclosed the perfect speaker. A good speaker should be capable of reproducing the entire audio frequency spectrum flat from 20 to 20,000 cps. Distortion other than frequency discrimination should be under 2 per cent throughout the range. It has been found that if a speaker reproduces the lows well, it cannot vibrate rapidly enough to reproduce the highs with sufficient magnitude. If the size of the speaker is reduced in order to enable it to vibrate at sufficient velocity and amplitude to reproduce the highs frequencies, the unit will lack low-frequency output because the speaker cannot couple to enough air to produce good low-frequency radiation. An 8- or 10-inch model is usually the best compromise.

Most of us dislike compromise and so we have found it advisable to use more than one speaker. Each speaker specializes in reproducing a specific portion of the spectrum. The obvious procedure is to connect all the speakers across the amplifier's output, and each will automatically reproduce its part of the spectrum. Division is automatic, since the speakers cannot reproduce each other's frequencies adequately. This method of connection is a poor one for at least three reasons, however:

(1) It is difficult to get good high-frequency response from even a small speaker unless all moving parts are extremely light. Therefore, many such units (tweeters) are structurally weak. If frequencies below those for which the unit was designed are allowed to enter it, their amplitude would be too great, the elastic limit of the mechanism would be exceeded, and of course, it would be destroyed. Even if the undesired frequencies did not ruin the mechanism, the output at these frequencies would be distorted because of the non-linear mode of vibration which the cone produces when confronted with frequencies which are beyond its capabilities.

(2) The low-frequency radiator (woofer) may have resonances in its upper register which are unpleasant. Therefore, it is best to restrict the range of this speaker so that the speaker will not be excited by energy at its resonant point.

(3) Each speaker placed across the line lowers the impedance presented to the amplifier. When each speaker is assigned a definite portion of the spectrum, this effect is minimized.

It is obvious now that we must divide the spectrum for use with specialized speakers. This division is accomplished through the use of circuits known as crossover networks, or frequency dividing networks. The discussion and circuits following illustrate the manner by which these networks accomplish their purpose. Notice that there is no sharp cutoff or transition at the point where one speaker takes over from the other, High-Q, sharp-cutoff filters would introduce ringing at each transition point, or crossover point as it is mostly commonly termed. 12 db/octave is usually the maximum gradient of attenuation above or below the crossover point, as the case may be. Some engineers advocate a more gradual slope. I personally favor slope between 3 and 6 db/octave. It must be remembered, however, that, with this gradual rolloff frequencies outside the range normally intended will be present in the various speakers comprising the system, and the speakers must be designed to handle more of these undesired frequencies than if they were used with a network having a higher degree of attenuation.

It sometimes happens that the woofer used with a particular speaker system is quite uniform in response up to, perhaps, 5000 cps, after which the highs roll off smoothly. There is, in this case, no need to limit the range the woofer is to handle. Obviously a tweeter must be used to restore the highs. It must not, however, operate to any great extent until 5000 cps is reached, after which point its output rises as that of the woofer falls. *Figure* 1 shows a circuit which could serve this purpose. C_1 is

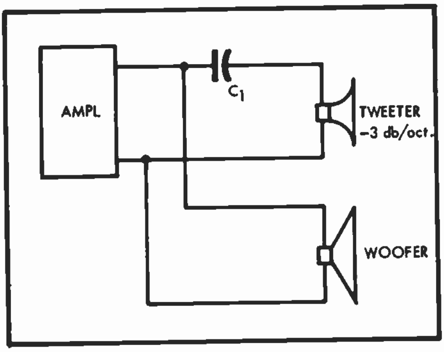

Fig. 1

chosen to have a reactance equal to that of the tweeter at 5000 cps. Below this frequency the capacitor has a reactance which is greater than that of the tweeter so more of the voltage is lost across the capacitor than is developed across the tweeter. As the frequency decreases, less and less power is available to the tweeter, since the reactance of the capacitor becomes larger and larger in comparison with that of the tweeter. As the frequency decreases, the tweeter is gradually isolated from the line. It therefore does not load the circuit when not in use.

Suppose the composition of the tweeter were such that, with the attenuation of-

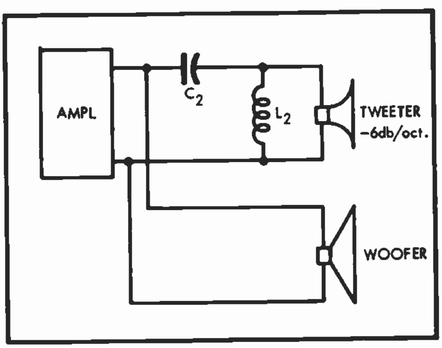

Fig. 2

fered by this simple network, the low frequencies were great enough to cause tweeter damage. The circuit shown in *Fig.* 2 can be used to cause a more rapid attenuation of the lows. This circuit is similar to that of *Fig.* 1, but has a shunted inductance. C_2 functions as before, but its value should be decreased to 0.707 times the value as calculated above.

L_2 is designed to have a reactance at the crossover frequency which is equal to 1.414 times the reactance of the tweeter at the crossover point. As the frequency decreases, the reactance of L_2 increases. This results in the total reactance to the capacitor being less than it would without the inductance across the tweeter, with the result that the voltage division between capacitor and tweeter is greater than the attenuation offered by the circuit of *Fig.* 1.

Suppose now that we have found it necessary to attenuate the high-frequency signals feeding the woofer. We still wish to attenuate the lows fed to the tweeter as before. The circuit of *Fig.* 3 could be used. Notice the similarity between *Figs.* 1 and 3. (Consider the tweeter hookup in *Fig.* 1, and the woofer hookup shown in *Fig.* 3.) In *Fig.* 3, L_1 is in series with the woofer, whereas in *Fig.* 1 C_1 is in series with the tweeter. L_1 is designed to equal the reactance of the woofer at the crossover frequency. As the frequency rises above this point, the reactance rises higher than that of the woofer, and again a voltage divider is formed. Less and less signal is available to the woofer as the frequency increases. As the tweeter takes over, the woofer is gradually removed from the circuit, "unloading" the line except for the tweeter. In all of the preceding circuits, the impedance across the line varies somewhat.

A circuit known as a constant-impedance network is shown in *Fig.* 4. This circuit attenuates the response of the woofer still further. Notice that the circuit for the woofer branch of the network is similar to that of the tweeter branch. Of course, the roles of the capacitors and inductances are reversed, since they behave oppositely with regard to frequency attenuation. This circuit is the standard parallel configuration of a constant-impedance two-way crossover network, and the impedance presented to the amplifier is essentially constant throughout the entire audio range. Values are as follows: $L_2 = 1.414 \times R_0/2\pi f$ and $C_2 = .707/2\pi f\ R_0$ where R_0 is the impedance of the speakers (and the output transformer tap) and f is the crossover frequency. Note that both inductances are of the same value and both capacitors are the same.

In many instances, it is desirable to add a third speaker to the system. The frequency range covered by the woofer is reduced to perhaps 250 cps. The third speaker covers the gap from 250 to 5000 cps, and is therefore known as the midrange speaker. *Figure* 5 shows the complete circuitry for a crossover network which can be used with a three-way speaker system. L_2 and C_2 are connected as in *Fig.* 4, and supply signal to the woofer. Also across the line from the amplifier are C_3 and L_3. Signals for the two remaining speakers are taken across L_3. C_3 is designed to attenuate all signals

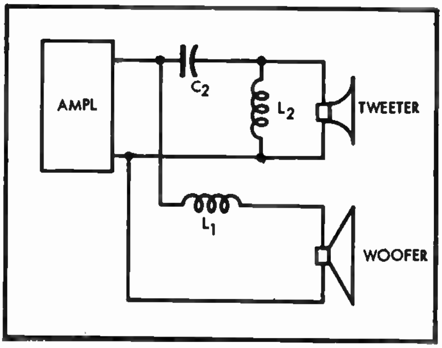

Fig. 3

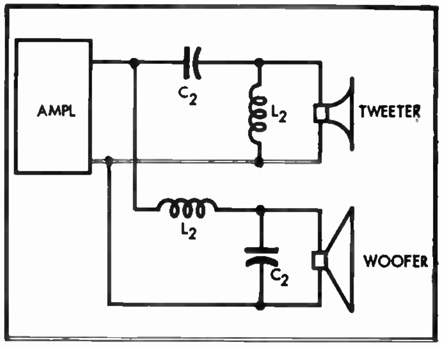

Fig. 4

below 250 cps, as is L_3. From L_3 we use another crossover circuit similar to that of *Fig.* 4. The remainder of *Fig.* 5 is, therefore, a two-way network. The midrange speaker can be considered to be the woofer, and the tweeter to be itself. Since attenuation below 250 cps has already been accomplished by L_3 and C_3, it is necessary only to attenuate frequencies above 5000 cps from appearing in the midrange speaker, and to attenuate those below this figure for the tweeter. Values for C_1, C_4, L_1 and L_4 are calculated as for C_2 and L_2 in *Fig.* 4, using a crossover frequency of 5000 cps, the point where the tweeter takes over. This wiring of the midrange is exactly the same as that of the woofer in *Fig.* 4, except that the values of its associated inductance and capacitor are different from those associated with the woofer. Like the midrange, the tweeter derives its signal across L_3. L_1 shunts the lows around the tweeter, while C_4 provides an easier path for highs than for lows. C_1 and C_4 have the same values, as do L_1 and L_4.

The frequencies chosen for purposes of this discussion are purely arbitrary. Some tweeters are made to take over at 7000 cps, while still others are designed to function at frequencies as low as 1000 cps. There is also considerable latitude with regard to woofers and midrange units.

The circuits presented here are basic, but there are variations. Some networks are designed to attenuate the response at even a more rapid rate than *Fig.* 5. Others are provided with switching provisions, so that a wide variety of crossover points can be obtained. Still others are series circuits; these are more costly to build, but many engineers favor them over the more common parallel constant-impedance networks.

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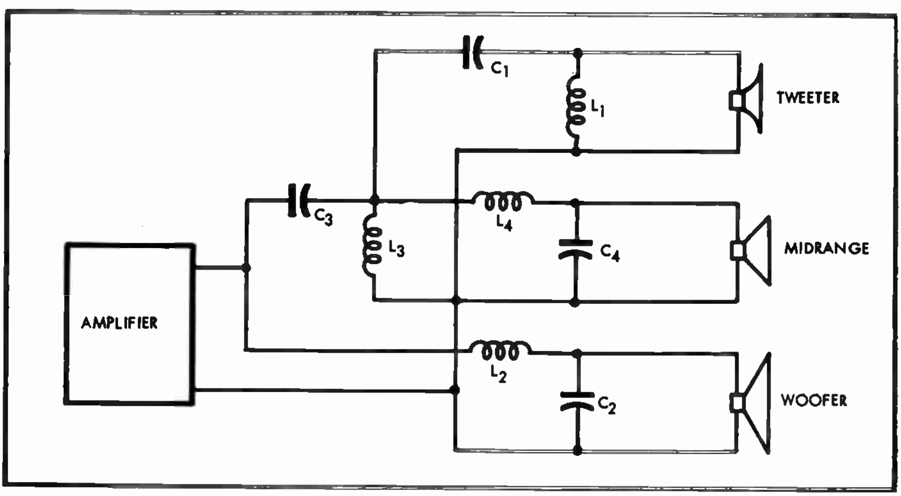

Fig. 5

Speaker Efficiency

Q. Would you please enlighten me as to the relationship between power output of an amplifier and the efficiency of a speaker? For example, I am using a speaker whose efficiency is 10 per cent. How powerful an amplifier do I need in order to drive the speaker sufficiently to obtain 20 watts of musical program level? Can it be that we only obtain two watts output from a 10 per cent efficient speaker when this speaker is driven by a 20-watt amplifier? Fernando Sim, Manila, Philippines.

A. When a speaker has a maximum power handling capacity of 20 watts, it means that 20 watts is the top amount of power which can be fed into the speaker without damaging it or causing serious distortion. If the speaker is 10 per cent efficient, it will when supplied with this maximum of 20 watts, produce 2 watts of acoustical output. Remember that, by definition, efficiency is equal to the power fed into a device, divided by the power output, or yield, of that device. 2 watts output, as in this example, sounds like a small amount of power, but bear in mind that a full symphony orchestra playing at top fortissimo only develops about a watt of acoustical power. Because of this, you will not need a 20-watt program level from your speaker system. (Such a level would most certainly injure your ears, were you to stand near the system when it is giving out with that much sound.) However, assuming that for some reason you do need such high program levels, and further assuming that your speaker system is 10 per cent efficient, you would need an amplifier capable of delivering a power output of 200 watts in order that these conditions be met. Further, you would need a speaker system whose power input capabilities are at last equal to 200 watts, and if this program level is to be maintained over long periods, you would need a system capable of peak power levels of from 250 to 300 watts.

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Speakers and Infinite Baffles

Q. We are considering the installation of an infinite baffle in a wall. Free cone resonance and efficiency are doubtless to be selected with some discretion in this regard. We understand that some of the best speakers for this use are in the relatively inefficient class. How is better bass definition attributed to low efficiency? What bearing has the flux density upon efficiency? What cone resonance will be best for most realistic reproduction in the home? L. B. Osborn, Newburgh, Indiana.

A. Efficiency is governed by the flux density and compliance. The less the compliance and the greater the flux density, the greater the efficiency of the speaker. Because the back wave is lost with infinite baffles, the cone must be free to travel a great distance to make up for this loss. Since low compliance and high flux density both hinder cone travel, speakers having these characteristics will not work well in an infinite baffle. If the cone is too compliant for the amount of rear loading upon it, the speaker may be damaged. For this reason, I like to use a speaker with rather high compliance and high flux density. The flux density limits cone travel after the cone is in high-amplitude motion and it thereby prevents possible damage during power peaks. Poor compliance is a constant restraining force and should be avoided. Unless your baffle is specially designed, I don't recommend a speaker having a very high compliance. You will have lots of intermodulation distortion at best, and you can probably damage the speaker mechanism.

Since an infinite baffle tends to raise the resonant frequency of the speaker cone, it is advisable to use a speaker whose resonance point is as low as possible. As the compliance increases, resonance is automatically lowered. If the infinite baffle is located in a small room, it is unnecessary to have a speaker with a resonant frequency much below 30 cps, since the volume of air contained in the room is not sufficient to reproduce frequencies much below 35 cps. A speaker's 30-cps resonance may be raised easily to 35 cps or higher, depending upon the size of the space behind the cone.

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Cabinet Dimensions

Q. I am thinking of building a large bass reflex cabinet of approximately these dimensions: 2½ ft.×2½ ft.×6 ft. I would like to know whether these dimensions are suitable, provided that the port is properly tuned. W. A. Long, Akron, Ohio.

A. In theory, the optimum ratios for the dimensions for a bass-reflex enclosure are 1:2:3. Therefore, assuming the largest dimension of your cabinet to be 6 ft., the others should be 2 ft. and 4 ft., respectively. Because of the large volume of such a cabinet, it will probably be necessary to scale down all dimensions, keeping their ratio intact.

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Infinite Baffles

Q. *Among the objections to the infinite baffle type of enclosure is that the pressure built up inside the cabinet by cone travel interferes with free operation of the speaker, primarily by raising its resonance. How about drilling one or two small holes in the side of the cabinet to relieve this pressure? At the same time, so little of the back wave would escape as to be negligible. Richard J. Galvin, Chicago, Ill.*

A. When a speaker whose suspension is flexible is used in conjunction with an infinite baffle, the speaker is in need of external damping. The pressure built up within the cabinet is a pneumatic spring which performs this damping function. Our own experiments have demonstrated that a speaker having a flexible suspension is the type which works best in such an enclosure, rather than one whose suspension is not very compliant. When the pressure is too low, the speaker will, among other things, tend to bottom more easily, possibly damaging it. Of course, if the pressure is too great the speaker will be over-damped and, therefore, impede the motion of the cone, especially at low frequencies, for it is on low frequencies that the cone travel is greatest. The easiest and most effective way to lower pressure is simply to increase the volume of the cabinet. If the box is large enough, no holes are needed to relieve pressure. If the box is too small, relieving the pressure will probably do little to improve performance, because the cabinet's physical size will not be sufficient to prevent front and rear wave cancellation.

Cable Lengths

Q. What are the maximum lengths of cable which may be attached to a loudspeaker of a given impedance without a loss of more than 0.1 db? J. Kass, Asbury Park, New Jersey

A. There are two variables which must be taken into account in order to answer this question. One is the impedance of the speaker and the other is the resistance of the connecting cable. For a given length of cable, it must be kept in mind that a cable is composed of two separate conductors, each of which contains a specific resistance per unit length. The resistance is inversely proportional to wire diameter. The following table shows the length of a cable used for connecting speakers to amplifiers with impedances ranging from 4 ohms to 500 ohms. The table represents the actual physical length of line, rather than the lengths of the two conductors laid end to end.

TABLE I

Wire Guage	Line Impedance in Ohms				
	4	8	16	150	600
22	20′	40′	80′	800′	3200′
20	30′	60′	120′	1000′	4000′
18	50′	100′	200′	1600′	—
16	80′	160′	320′	2400′	—

Distance from Crossover Networks

Q. *My question is: How far can speakers be operated from their crossover networks, assuming that fairly large wire is used? If the distance over which this operation is permitted is considerable, say 40 feet or more, then why should one not use the same crossover network for a number of speaker combinations? W. E. Dancey, Houston, Texas.*

A. In this column, Jan., 1958, there is a table showing lengths of line *vs.* wire diameter, *vs.* impedance for 0.1-db loss. By using this table as a guide, negligible distortion of the characteristics of the crossover network will result. This is because the resistance of the line is negligible as compared to the nominal impedance of the circuits involved.

One word of caution: If the crossover network is of the non-adjustable type, be sure that all the speaker systems to be used require the same crossover point. If the speakers are to be switched (and this is certainly the best way of handling this project), be sure that the switch contains sufficient positions, one for each speaker system, and enough poles, one for each element of a speaker system.

Balance

Q. I have a fine amplifier in most respects, but I do notice that, as the gain is reduced, the balance between highs and lows is not of the same proportion as when the gain is turned up. There seems to be a very definite point at which the frequency becomes good. What is this condition and how can it be corrected? Name Witheld, Lake Worth, Fla.

A. This sounds as though it might be an advanced case of Fletcher-Munson trouble, and the effect is due to the lowered sensitivity of the human ear to low frequencies as level is reduced. On a perfectly flat reproducing system, music will appear to have the proper balance only when the reproduction level is identical with that of performance. If this is the problem in your case, you might find a solution in installing a compensated type of volume control in place of the "flat" or uncompensated control now in use. There are other causes which could result in a loss of highs as the volume is lowered, the most common of which is due to high stray capacitance in a grid circuit. As capacitive reactance at the higher audio frequencies approaches the input impedance of the amplifier, the amplitude of the high frequencies is diminished with respect to middle and lower frequencies. One form of such capacitance is that between the resistance element of a volume control and its case. This varies with the setting of the control, so that when the control is in the most advanced position, the effects of the shunt capacitance are at a minimum. The larger the resistance of the control, the smaller will be the shunt capacitance needed to cause this high-frequency loss. To minimize this effect, replace the present volume control with one whose resistance is no more than half that of your present control. By whatever amount this resistance has been decreased, the value of the coupling capacitor feeding this control should be increased, so as not to impair the low-frequency performance of the amplifier. If the value of the coupling capacitor is not increased to meet the decreased control resistance, losses will occur because of the voltage divider formed by the reactance of the capacitor and the resistance of the control. As frequency decreases, the reactance of the capacitor increases until finally there comes a point where that of the capacitor equals that of the volume control. As frequency proceeds below this point, it is obvious that more voltage will be lost across the capacitor than across the control. From this point on, low frequencies will be attenuated at the rate of 6 db per octave. Æ

Power Output

Q. I have been checking the power output of my Williamson-type amplifier and have had some disappointing results. I am using 6L6-GB tubes as power output stages, driving a UTC LS63 output transformer, right into an 8-ohm speaker. The tube manual states that these tubes, operated with cathode bias with 360 volts on their plates and 270 volts on their screens, would deliver 24.5 watts in class AB. I'm operating the tubes at 340 volts on the plates and 250 volts on the screens, and with a 250-ohm cathode resistor, bypassed with 250 μf. The grids are wired with 470,000-ohm resistors. If I disconnect the driver stage from the 6L6 grids and feed it into separate 470,000-ohm resistors, I can obtain 15–16 volts RMS to drive the grids of the 6L6's. This waveform is clean as seen on a 'scope. However, as soon as I connect the driver to the 6L6 grids can only drive them to 10 volts RMS. Beyond this point, the waveform flattens off. At this point I can get only about 10 watts out. Can you explain why I cannot approach at least the 20-watt level with the above parameters? W. E. Henry, Albuquerque, New Mexico.

A. It is logical to expect that, when the output of your driver stage is connected to a dummy load, more output would be obtained than when the same driver was connected to its proper grid load, with the remainder of the circuit made operative. This is because the output stage is designed to feed back a certain amount of its energy in such a direction as to cancel a portion of the driver's output voltage. A driver stage of higher capacity would have to be used to overcome this difficulty. Such a stage might take the form of a push-pull parallel configuration.

Think back to the original Williamson circuit, in which 807's were used. In some circuit arrangements as these tubes are easily capable of delivering 40–50 watts output, and hams use them as Class B linear amplifiers and easily obtain 100 watts from them. Williamson got 8 watts out, using them as triodes, but with very low distortion. Æ

EQUIPMENT

PROFILE

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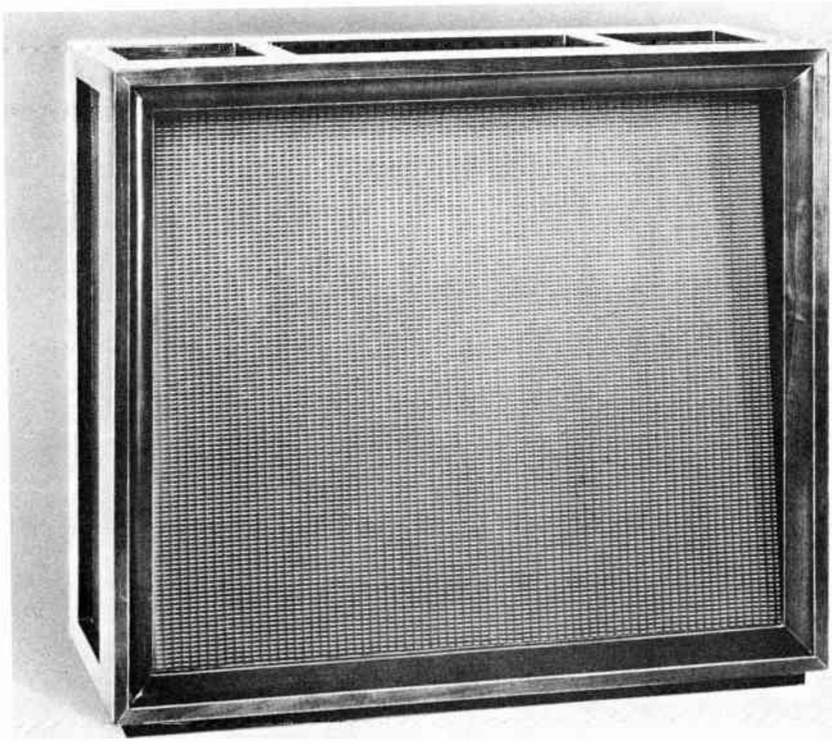


"Windsor" de luxe model

WHARFEDALE THREE-WAY SPEAKER SYSTEMS

These two speaker systems represent a new approach to ''enclosure'' construction, since they are only enclosures in the sense that they provide a housing on which the speakers can be mounted—the backs of both models are open.

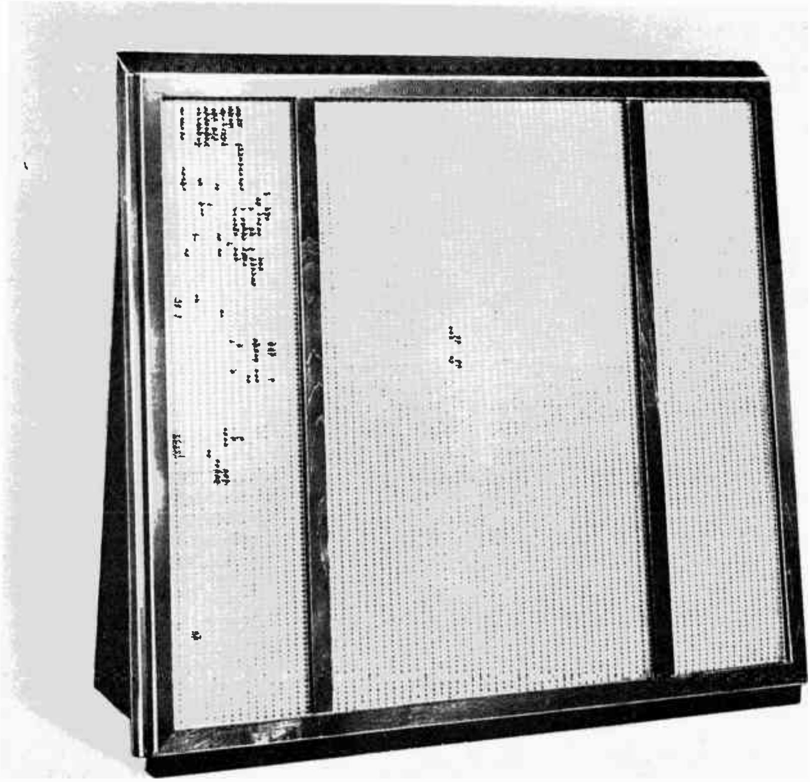
Designed by Gilbert A. Briggs, these models are relatively inexpensive when one considers the quality of performance. Three speakers are mounted in these cabinets, a 12-inch unit at the center near the bottom, a 10-inch above and to the left, and a 3-inch tweeter to the right of the 10-inch speakers. In the Warwick model—the less expensive—the 3-inch cone faces forward, while in the Warwick it faces upward. In essence, the Windsor, *Fig.* 4, consists of the Warwick, *Fig.* 3, with a rectangular framing around it for the sake of appearance. Both employ a sand-filled baffle which damps all vibration very effectively. And with no cabinet to speak of, there can be no ''cabinet resonance.''

The speaker and the housing were designed for one another, and neither speakers nor the baffle are available separately. The speakers employ plastic foam suspension, cast baskets, and the usually high flux density encountered in all Wharfedale speakers. No dividing networks—which are claimed by some to introduce their own forms of resonance—are used, the low frequencies being excluded from the tweeter by a series capacitor. The bass resonance on the Windsor model observed was measured at 33 cps, and while no absolute output measurements were made, output up to 21,000 was definitely measureable.

The principal quality of this system is in its pleasant listenability. At first it might appear to offer nothing unusual in the way of quality, but it is the sort of sound reproduction which grows on one. It is never strident, and for this reason does not make the flamboyant impression on the listener that is usually encountered in an audio show. On the other hand, we have long insisted that while audio shows may be criticized for the quality of sound —along with the general loudness level— people just won't come into a room unless the sound is on the spectacular side. And we have always been willing to bet that the listener who takes home on the the more spectacular speakers will tire of the super-highs and super-lows quality within a few hours of listening.

Such is not the case with either of these models. When the speaker was first received, it was connected up to the office system, and—except for being taken out for a weekend to compare it in better known listening areas—it has been running practically all day long every day. It is for this reason that we can unqualifiedly recommend these models for their excellent listening quality—particularly for those who want to get away from the point-source quality of sound. Neither of these models is ever likely to be accused of causing "listening fatigue"—that elusive, but often present characteristic of some loudspeaker systems.

One of the unusual features of the reproduction quality of these speakers is that they are relatively unaffected by their position in the room, and the sound output appears to be about the same whether they are against a wall, in a corner, or free standing. Because of this quality, they are especially suited for use in pairs for a stereo system.



The less expensive "Warwick" model. Both employ the same Components mounted on a sand-filled baffle, and both are open at the back.

KLH Research and Development Corporation loudspeakers

WITH EACH NEW loudspeaker system that is introduced on the market, we seem to become more and more convinced that the last word in speaker design has not yet been reached. After many years of listening to practically all of the speakers and enclosures that have ever appeared, we have learned at least one thing—one cannot judge a speaker and/or enclosure in just a few minutes of listening. One must live with a speaker long enough to give it a very wide range of program material to be able to make a valid judgment. Because of this, it would seem that some enterprising dealers would make a "sale" of a loudspeaker—type and brand name unspecified—and then allow the customer to try out three or four of them at home (all in the price range selected by the customer himself) before being finally commited to a particular choice.

We have had the privilege of "living with" the largest unit in the relatively new KLH line of loudspeakers and we have heard the latest model compared directly to the big one. Then we made a trip to Cambridge to see something of the way the units are built. By this time we feel that we are competent to judge them intelligently.

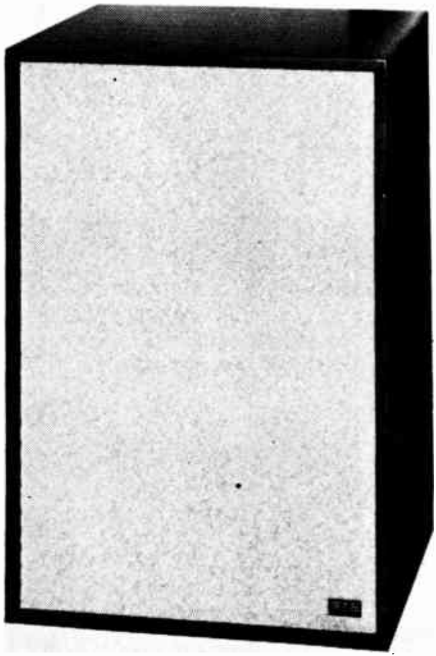


Fig. 1. KLH Model One low-frequency speaker which provides room inside at top to accommodate high-frequency units such as the KLH Model Five.

Model One combined with Model Five—KLH uses spelled-out model numbers—is by no means an inexpensive loudspeaker, falling rather into the top price bracket. But with extended listening, one can only come up with the description that this system is a "loudspeaker that does not sound like a loudspeaker." The cabinet, shown in *Fig.* 1, is 38 in. high, 25 in. wide, and 16 in. deep. It consists of two 12 in. speaker mechanisms enclosed in separate air-tight 2.25 cu. ft. enclosures in the lower part, and a space 7½ in. high by 23½ in. wide and 14½ in. deep at the top to accommodate a high-frequency speaker. When supplied as "Model One and Model Five," the space is occupied by the KLH high-frequency speaker system—the Model Five part of the name—but the space is adequate to accommodate a JansZen electrostatic high-frequency speaker if desired. As Model One alone, the upper portion is open in the furniture cabinets—mahogany, birch, or walnut finishes being normally available. The Utility model is finished in dull black and is 27½ in. high over-all—no space is provided to the high-frequency speakers.

Model Five consists of a box in which are mounted three small direct-radiator cone speakers—the one in the center covering the mid range, while the two outside ones, splayed out slightly, cover the high range. Switches on the back allow for operating the two sections separately at normal, below normal, or above normal settings, which gives the user flexibility in adjustment for specific acoustic conditions in his listening room.

Extended listening to this One/Five combination gives the impression of listening to the original performance rather than to a loudspeaker. Listening without looking—a desirable practice when comparing speakers—one feels that he is hearing a direct performance. The effect is almost uncanny, because everyone is conditioned to previous experience somewhat, even though aural memory is not actually very long. This combination of speakers sounds exceptionally smooth, and no coloration seems to be apparent. Of course, this was the main objective of the manufacturers, but they have achieved this objective quite well.

In direct listening comparison to the "Standard" speaker, we would be inclined to believe that the KLH had less bass—even though it has good output without doubling down to about 27 cps while the Standard goes to about 24. There is some accentuation of the very low frequencies in the Standard system, which makes it extremely listenable at low levels, but this almost seems too much after listening for a half hour or so to the One/Five. At its price, however, it *should* be a good loudspeaker.

The New Model Six

In March, the new Model Six was introduced to the press and trade. It is considerably smaller than the One, being only 23½ in. high by 12¼ in. wide and 12⅝ in. deep. By itself it is capable of a performance which we have come to think of as unbelievable for so small a cabinet. Two in a stereo pair are excellent.

On the Cambridge trip, we asked to hear the Six in direct comparison to the One/Five—a rugged demonstration any way you look at it, for the Six costs only about one-fourth as much as the One/Five. We came out of that test with the comment that we should hate to have to convince the better half that we should spend four times as much for the bigger model. There was a difference, to be sure, but it was not as apparent as one would expect and on some material, in fact, the unit that was 1½ db louder sounded best, regardless of which one it was. (We were unable to match levels exactly—the Six could be varied by 3 db, and it was either above or below the One/Five.)

The reasons for this similarity in sound quality may well be in the construction of the Six. The entire frame, baffle, and magnet-supporting structure are bonded together forever in a fiberglass-epoxy compound, making the speaker and baffle essentially one single unit. The high-frequency unit is similarly set into the baffle, although it is almost entirely self-supporting in the magnet structure, and is fitted into the baffle in that manner to make sure of air-tight construction. The board itself is sealed in the cabinet, with the volume practically filled up with rock wool "blocks."

KLH is one of the very few U. S. manufacturers to make their own cones—Bozak is the only other one we know of. We have seen evidence of much experimentation in the "junk box" where several hundred tried-and-discarded cone types have accumulated, in both woofer and tweeter types. The cone for the woofer is relatively heavy, and—because of its tar content—might be likened to a piece of roofing paper. It is also quite stiff, and is not likely to break up over the range in which it works. The tweeter cone is light and hard, much like a parchment bond paper. On the whole, the KLH line shows the results which can be obtained by intelligent and persistent research.

One demonstration we saw was of particular interest because it showed how smooth the tweeter is. Using machine-run curves and a calibrated Western Electric 640AA condenser microphone, several different types of tweeters were compared. The effects of diffraction around various baffles were readily noticed, and the curve of the small unit used in the Six was considerably smoother than we have come to expect from cone tweeters, with response extending out to well over 15,000 cps. We still prefer the One/Five, naturally, but we would be quite well satisfied with the Six—particularly if we needed two of them for a stereo pair.

Fig. 2. The new Model Six is only 23½ in. high but sounds much bigger.

Fig. 1. The Pickering "Isophase" electrostatic loudspeaker.

PICKERING ISOPHASE LOUDSPEAKER

The first time we heard this model—at the 1955 audio show—we were considerably impressed with the quality of reproduction, which we described in the succeeding editorial as being like "a window on the studio." Like any new item, this speaker had both proponents and opponents, and during the past two years it has undergone several improvements, as would be expected, so that today it is a superb device. We have had the pleasure of living with the newest model for over a month, and we still believe that the quality of reproduction is about all that could be desired. In short, we are most enthusiastic about it.

Basically, of course, the electrostatic speaker is not a new device—one was on the market commercially in a console radio as far back as 1932 under the name "Peerless 'Kylectron'" and was, for its time, a fairly good loudspeaker. We had a friend who owned one, and upon its periodic puncturing of the diaphragm he would go to the corner drug store and buy a small device manufactured by Klienerts and some chocolate bars manufactured by Hershey—those that were wrapped with very thin tinfoil. He would then cement the foil onto the rubber, replace the assembly into the speaker unit, and everything was fine again until another puncture.

As we recall it, this speaker was single-ended—that is, it did not employ an electrode on each side of the diaphragm. Modern electrostatics are push-pull, and the diaphragm is spaced between the two outside electrodes, as seen in *Fig.* 2. Advances in plastics make it possible to utilize a very thin diaphragm on which a metallic coating has been evaporated, resulting in greatly reduced mass, so that the spacing between the outer electrodes and the diaphragm is maintained by myriads of tiny "stand-off insulators," the electrodes being not unlike a flocked screen. The over-all area of the speaker diaphragm is 730 square inches, and the complete assembly is 36 inches wide, 28 inches high, and 8½ inches deep.

Because of the large area—even at a practical spacing—the speaker has a capacitance of approximately .0025 uf, which is reflected at the input terminals of the divider as approximately 12.25 µf. This type of load can be troublesome with poorly designed amplifiers, but we have tried it with the 70-watt Heathkit described on the following page, with both Dynakit II and Dynakit III, with the Marantz power amplifier, and with a 65-watt Fairchild amplifier and performance has been satisfactory with all of these.

Earlier models of the Pickering Isophase were thought to be inefficient, and they were often coupled with inefficient woofers because of this. Present production has overcome this difficulty, and we are currently using the unit with the woofer of a United Speaker Systems' "Premiere"—which is an Altec 803, and of recognized high efficiency. The Pickering 401E divider, which also furnishes the polarizing voltage for the electrostatic unit, is equipped with an attenuator in the woofer circuit, but it is properly balanced with the attenuator at its maximum position, so the Isophase is essentially of the same efficiency as the Altec woofer.

The Isophase has a frequency range from 200 to 35,000 cps, and will accept the full power output of a 50-watt amplifier at 8 ohms without damaging the diaphragm, but we have never found it necessary to turn the volume up to that extent—and we like fairly high listening levels.

One feature we consider excellent is the provision of a phasing switch on the divider which permits reversing the relative phase of woofer and tweeter. This is much simpler than having to reverse the woofer leads physically at the terminal strip, and permits making rapid checks to determine the correct phasing position. Once set, of course, there is no need for change. The Isophase speaker and its associated 401E divider are designed to work from an 8-ohm amplifier tap. The crossover frequency is 500 cps, which is fixed in the divider. The divider utilizes a 1V2 rectifier tube to furnish the 1500-volt d.c. for polarizing, and the power requirement for the divider is only 10 watts so we find it simpler to leave the unit plugged into the a.c. line all the time, since the speaker is some distance from the amplifier and running an additional a.c. line is not convenient.

Performance

From the subjective standpoint, we have noted things on records which we had never heard before. With no claim for any stereophonic effect from the Isophase, it does appear to have a spacial quality which is hard to describe. As one moves around within the 55-degree pattern of the speaker, the sound seems to originate at varying parts of the diaphragm. Moving up close to the unit, with the ear practically against the grill cloth, one notes that the actual level does not appear to be very great. This is logical since the ear hears only a small portion of the diaphragm, and the small excursion of the diaphragm moves only a small amount of air at any given point. Because of the large surface, however, the over-all sound output is comparable with any other type of loudspeaker. We would not say that good performance is not attainable from conventional types of tweeters—we have been quite content with them for years. But if enough space is available for an Isophase, it is quite probable that the quality of reproduction may bring a new pleasure to listening.

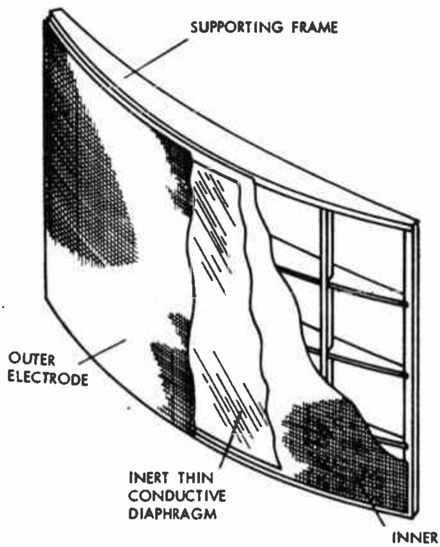


Fig. 2. Diagram of internal construction of the Isophase.

BAKERS ULTRA 12" LOUDSPEAKER

The performance of this 12-inch speaker is rather exceptional, from its low resonance to its high usable frequency of output. Free-air resonance of the one sample we measured was at 28 cps, with the resonance peak measuring only 32 ohms for a nominal impedance of 15 ohms at 400 cps—due, we imagine, to the use of a high-strength magnet and a voice coil wound on an aluminum former. The cone surround is a thin membrane of plastic foam, which permits large cone excursions, and the voice coil is sufficiently long so that it extends more than ¼ in. on each side of the magnetic gap, thus ensuring the gap being filled with the same number of turns at all times, even at high excursions of the cone. This is claimed to account for the low distortion of the speaker, which seems to be reasonable.

The frame of the unit, *Fig.* 2 is cast aluminum, and is sufficiently rigid that it is not likely to be deformed by unequal tightening of the mounting screws. Flux density is said to be 18,000 gauss, which is high for a 12-inch speaker. Efficiency is somewhat less than average, but that is noticeable only in direct A-B testing.

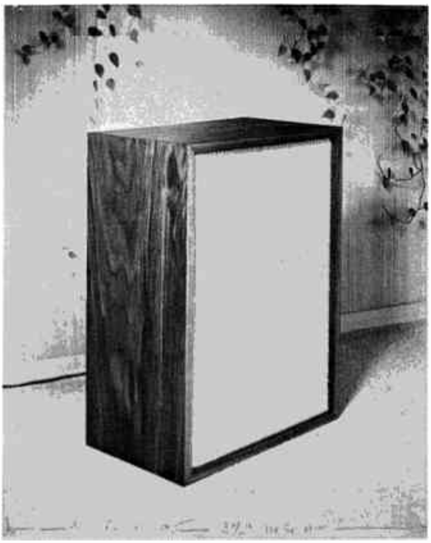
This unit was checked in a Bradford Baffle—only slightly larger internally than the speaker iself. This enclosure features an opening in the rear which is kept closed by a swinging plate. Mounted on ball bearings, this plate can accommodate internal pressures so that no padding is required in the enclosure itself, and no special adjusting of port area is required for different speakers. Having no previous experience with this baffle, we measured the voice-coil impedance in the box as well as in open air. While the open-air curve showed about a 2 to 1 variation in impedance over the resonance peak area, the impedance variation of the speaker in the Bradford Baffle was something less than 10 per cent. We consider this to be rather remarkable, since most enclosures tend to raise the resonance frequency as well as boost the amplitude of the peak. While the theory of this type of enclosure may be beyond most of us, it must be said that the Bradford Baffle—at least with this speaker—does what enclosures are supposed to do.

TANNOY "BELVEDERE" SPEAKER SYSTEM

Employing the well-known dual-concentric Tannoy speaker, the "Belvedere" is an innovation in this company's cabinetry, since it is considerably smaller than anything they have shown to date. But with the U. S. market becoming increasingly conscious of size—with a consequent demand for speaker systems which, when used in pairs, still leave some space in the listening room for people, chairs, and perhaps an equipment cabinet—some steps in this direction were necessary.

The "Belvedere" is of intermediate size —measuring 26 in. high by 18 in. wide and 12 in. deep, or if the user prefers, 18 in. high by 26 in. wide, since all four surfaces of the cabinet are finished so the unit may be used in any position. The cabinet is of exceptionally heavy construction, and is a modified bass-reflex model. In addition to the opening for the speaker cone, there are two additional openings as ports, approximately 1½ x 14 in. each, on either side of the speaker. The interior of the cabinet is lined with acoustic material, and a controlled thickness of the material completely covers the ports, resulting in a considerably higher loading on the cone at the low frequencies.

Over the period of development of this enclosure, Tannoy engineers ranged from an infinite baffle to completely open ports, both with and without acoustic treatment in the interior—the cabinet size being fixed at the beginning as one which market surveys had shown to be ideally acceptable for the home. As an infinite baffle device, the low frequencies were somewhat attenuated, although they were clean and completely free of "muddiness." At the



other extreme—unloaded ports—there was adequate bass, but a noticeable muddiness. The design finally accepted retains the good features of both, and the resistance loading of the ports allows excellent bass response with a minimum of coloration.

Using the organ test record again, the speaker may be said to perform properly from D of the second octave, or possibly even down to the second C, which is 32 cps. We would describe the lower limit unquestionably as 35 cps, or perhaps even a cycle or so lower. At the top end, the output is a function of the tweeter section of the Dual-Concentric Tannoy speaker, and this is measurably flat up to 20,000 cps, although we do not hear that high.

Tannoy speakers are noted for their ruggedness, and in properly designed cabinets they cease to sound like loudspeakers but seem to sound more like the original instruments. There is a crispness throughout the voice range which gives a solid feeling of clarity to speech, yet there is never any "chestiness" which indicates a peak in the 150- to 300-cps region. On the whole, Tannoy speakers are remarkably lifelike, but we had never before heard them in small cabinets. This particular design, however, retains the high quality of performance that we first noted in the G. R. F. "Autograph" models when we first saw and heard them.

This is one of the models that we would not hesitate to describe as having made no compromises between size and performance—the performance is still there even in the smaller enclosure. It is interesting to consider how much speaker development has been done in the past year or so to reduce cabinet cost as well as size, and it is likely that more development will follow so as to result in a completely satisfactory stereo speaker system in not more than one cubic foot of space, yet retaining the spatial characteristics needed for good stereo reproduction.

UNITED SPEAKER SYSTEMS' MODEL X-100

Back in September, 1956, after comparing the United "Premiere" speaker system with our standard system, we wrote, "On the whole, however, we consider this to be one of the very few that we have heard that compares so closely to the comparison speaker." The old Premiere, with its 800-cps crossover and high-frequency horn system, has been improved—even though the improvement is only noticeable when direct comparison is made with the older model—by the change to a 500-cps crossover and the necessary change to a larger horn. The improvement is not glaring by a long ways, but after careful lisening is must be admitted that there is a slight improvement in the midrange with the Premiere 500. All of this is beside the point, however, since we are primarily discussing the new and considerably less expensive X-100, shown in *Fig.* 2. This model shows the importance of good enclosure design in getting good performance out of medium-priced components, for it is perfectly obvious that an Altec 803A woofer and an Altec 802A driver with the necessary high-frequency horn can not be furnished at the price of the X-100.

However, we are inclined to say that—like much high-fidelity equipment—the differences in performance are rarely noticed *on average program material* during a typical listening session. Of course there is a difference between a $300 ultra-super-special FM tuner and the economy model at $19.95 (if there is such). But on a high percentage of broadcast material the difference might not be noticed. In the same way, there is a difference between the X-100 and the Premiere, but on 75 per cent of the program material the difference might not be noticed.

Using our customary test record for low-frequency performance, the X-100 was observed to be a good performer from about 37 cps up, with the upper limit beyond our own hearing range. The response is smooth throughout the entire range, and the coloration—if there is any—is similar to that of the Premiere, which in turn is similar to our Standard system. On a subjective basis, which is the only way we feel that speaker testing can be done without anechoic chambers and fabulously expensive measuring equipment (and who listens to speakers in an anechoic chamber anyway?), we believe the X-100 to be an exceptionally acceptable speaker system.

Supplied in 8-ohm impedance only, the X-100 is available in African mahogany, Swedish birch, or pewter walnut. It measures 24 in. high by 24 in. wide and 15½ in. deep, and a cabinet of identical appearance is available for housing phono and amplifier equipment with a designation of X-100-E.

Technically, the X-100 consists of two 12-in. woofers and a single cone tweeter, with adequate cabinet damping for excellent transient response. Efficiency is relatively high for small speaker enclosures, with ¼ watt being sufficient to drive the speaker at full room volume.

Fig. 2. United Speaker Systems' new Model X-100.

HARTLEY 217-DUO STEREO SPEAKER SYSTEM

Under most circumstances we would not consider that a single cabinet only 36 inches wide could suffice as a complete stereo system, in spite of the fact that we have suggested previously a modification to a conventional corner cabinet for stereo application in a small room. But when one considers that practically all listening is done in typical living rooms instead of in anechoic chambers, one must realize that reflections from walls and furniture have a large effect on the sound pattern in a room and thus modify the classic characteristic which might be deduced from two sound sources spaced some six to eight feet apart. Obviously, of course, if one were to listen to the 217-Duo in the middle of a prairie it is doubtful if much stereo effect would be observed. In the average room, however, it is more than adequate.

The 217-Duo, shown in *Fig.* 3, is 36 in. wide, 30 in. high, and 15 in. deep, and houses two Hartley 217 full-range speakers splayed out about 30 deg. from the center line between them. When used as a monophonic system it shows a pleasant wide-source effect, completely free of the oft-described "hole in the wall" feeling. As a stereo speaker, under direct A-B listening tests with two conventional speakers spaced 8 feet apart, the single cabinet with the two splayed speakers gave a better over-all sound, and the stereo effect was distributed throughout the entire room so that no matter where you listened the stereo spread was still there. We believe there is much work yet to be done to determine just what is the optimum speaker for stereo, and we have learned that if there are two speakers in the room, the listener is likely to hear two speakers as separate units, rather than as parts of the whole—which is the principal reason why this observer insists on evaluating stereo installations with both eyes covered, and this applies equally well to a two-way monophonic speaker system when both speaker units can be seen.

Be that as it may, the 217-Duo does have a better stereo effect with the eyes closed—as does any other system in which two speakers can be seen. We have suggested that the grille cloth cover the entire front in one sweep.

As to the actual quality of the 217-Duo, we found it capable of going down to below 40 cps, and to have considerable audible output at 14,000 cps—above which we do not think we can hear very much, nor do we believe much source material extends that high, even if the records could retain it or the pickups all play it. Quality was judged by many listeners as excellent, being described by the more experienced as smooth and free from objectionable peaks—purely subjective, to be sure, but it is fairly well established that the choice of a loudspeaker is pretty much a subjective thing anyhow. Let it suffice that one compare speakers for himself, preferably on the same material and in the same acoustic environment, rather than accepting the judgment of some one else. But to these standards of judging, we can only say that we would consider this speaker to give good quality and an excellent stereo effect in any room larger than 9 x 12.

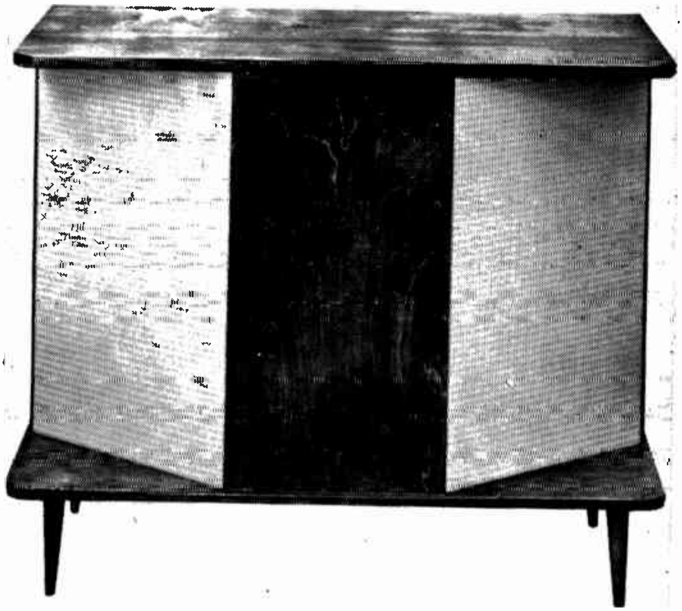


Fig. 3. Hartley 217-Duo—a complete stereo system in one cabinet.

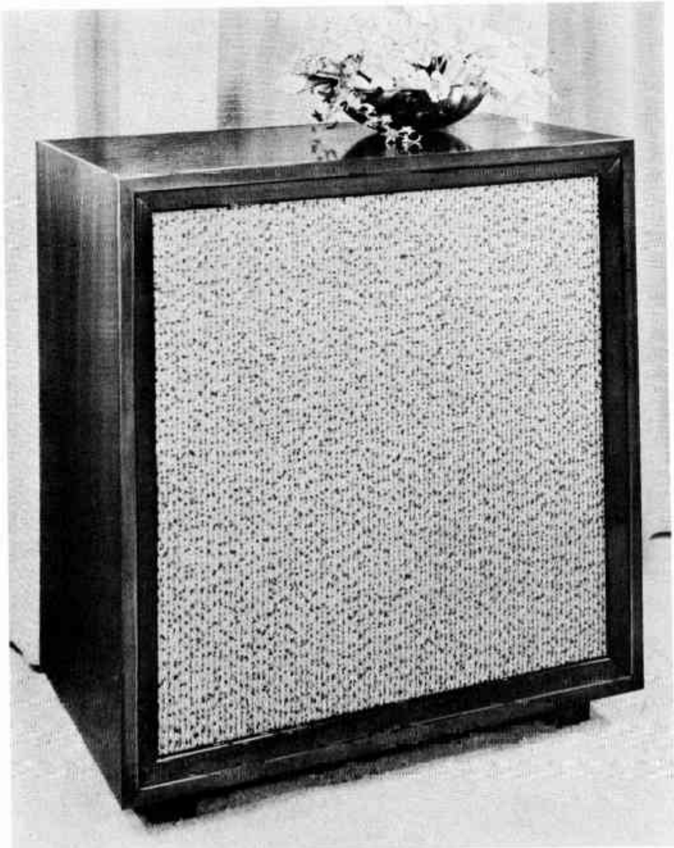


Fig. 1. Audio-Tech Laboratories' speaker system.

AUDIO-TECH LABORATORIES SPEAKER SYSTEM

Designed by Joseph Giovanelli, who conducts the AUDIOCLINIC column, the Audio-Tech speaker system offers some unusual characteristics at a reasonable price, and while smaller than many of the more familiar units on the market, its performance puts it alongside many of them. The unit, roughly 30 by 30 by 15 inches in size, employs a 15-inch woofer together with a tweeter to cover the range from 30 to 17,000 cps, according to the manufacturer. The woofer is built to the designer's specifications, and is suspended with an elastic material which provides extremely high compliance. With adequate acoustic damping and a very heavy cabinet, bass response is smooth and without apparent resonances, with the exception of a measurable increase in impedance at 50 cps. The nominal impedance of the system is 8 ohms, with the maximum impedance throughout the entire audio spectrum not exceeding 20 ohms—which indicates an almost complete lack of audio resonances. Because of the tweeter coupling circuit, there is also a peak in the impedance curve at around 2000 cps, above which the impedance drops to the nominal value up to 8000 cps, when it begins to rise again. The over-all result of the smooth impedance curve is reflected in a smooth mid-range. The crossover network is of the L-C type, with the woofer rolling off naturally to the 5000-cps crossover point.

In listening tests, the speaker was found to perform down to about 32 or 34 cps, using the "King of Instruments, Vol. I" organ record for a quick determination. The scale played on side 2 of this record begins at 16 cps, and as the tones go up the scale, the listener can determine easily the note at which the tones begin to sound musical. From 16 cps up to that audible change in sound quality, the tones come out as low-frequency flutters or wheezes, with no musical quality whatever. Thus by counting up the scale as the record plays, the listener can recognize immediately the lowest frequency at which the speaker begins to function as a speaker should.

Our second test record—for a quick analysis of speaker performance—is the old recording of Varese's "Ionization," (Elaine Music Shop, EMS-401), a record definitely not recommended for speaker testing in early morning hours. If we had to choose a single record to indicate speaker performance, this would be the one, for it is full of transients which serve to show high-end performance, while the eight-foot bass drum shows the low-frequency performance. It is not satisfactory as a quantitative measure, of course, but for a quick qualitative analysis it serves very well. The test oscillator is more useful for a thorough study, to be sure, but these two records are available to anyone and will give plenty of information.

The Audio-Tech speaker gives an excellent account of itself on the high-frequency end of the spectrum, both with the oscillator and with the test record. Transient performance is excellent, and output is readily audible up to this observer's limit, which is about 14,500 cps.

In addition to its good listening quality, the speaker is equipped with two refinements which all speakers should have—one is a calibrated control for tweeter balance, and the other is the use of colored binding posts for terminals to indicate polarity, thus making it easy to connect in proper phase relation. While we prefer a screwdriver-set control for balance, so as to avoid tampering or unauthorized changing of the setting, the calibration provides some reference for resetting the control to its "normal." We believe a speaker system should be set up once for its acoustic environment and then left alone permanently, any necessary changes in tone being made thereafter with the usual bass and treble controls.

The Audio-Tech speaker is available in bleached and dark mahogany as standard, and in other woods at a slightly increased cost. The cabinet design is simple, as seen in *Fig.* 1, and should fit into practically any modern home.

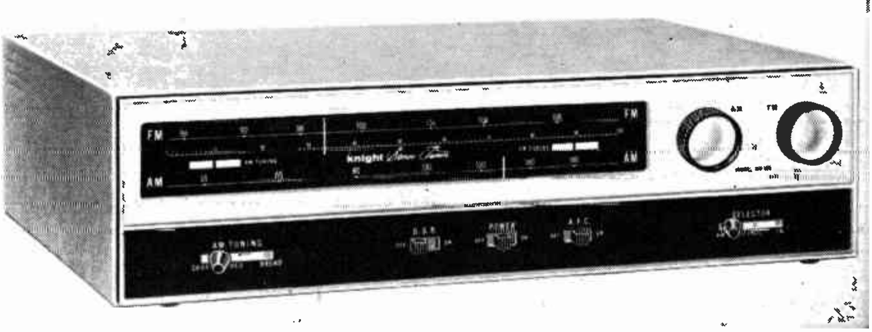


Fig. 9. Allied Radio Corporation's Knight KN-120 Deluxe AM/FM/Stereo tuner.

KNIGHT KN-120 DELUXE BASIC AM/FM/STEREO TUNER

New tuners seem to be introduced at the rate of about one each month, and aside from variations in appearance, quality of construction, and placement of controls, there is little to distinguish one from another. Performance is uniformly good, and while there are differences in sound quality, hum level, drift, and sensitivity, most of them use essentially the same circuits and many buy their i.f. transformers from the same source. But every so often something that is sufficiently new to warrant attention appears, and the Knight KN-120 Deluxe basic AM/FM/Stereo tuner is one of them. For in this model there is a radically new idea which we have never before seen applied to tuners, although all of us are familiar with the use of feedback in amplifiers—in fact, we rarely see an amplifier nowadays without feedback.

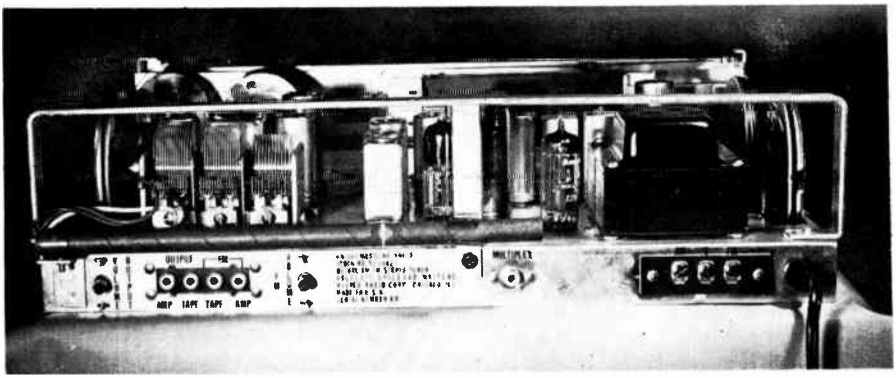


Fig. 10. Rear view of the Knight basic tuner.

Before discussion of the physical and performance characteristics of the KN-120, the application of feedback to a tuner should be explained, for it is in this area of circuit design that this tuner is distinguished.

Reviewing briefly, most readers are aware that a.f.c. may be obtained by connecting—through suitable filtering circuits—the output of the discriminator to a reactance tube which controls the frequency of the h.f. oscillator in the "front end." The Foster-Seeley discriminator is so arranged that when properly adjusted its audio output is superimposed on a d.c. voltage whose average value varies as the center frequency of the i.f. channel is varied. When tuned exactly "on the nose," the average voltage of the discriminator output is zero; if the set is tuned to one side or other of exact center—which means that the carrier is not distributed symmetrically over the straight-line portion of the discriminator curve—a d.c. voltage is developed. A reactance tube is a voltage-sensitive device which varies its basic characteristic—either inductance or capacitance, depending on the actual circuit—as the voltage applied to one of its elements is varied. If the tube is connected across an inductance in the oscillator circuit, the frequency of the oscillator can be varied by varying the control voltage. Thus the entire circuit functions like an engine with a governor in which a speed-proportional control is exerted on the engine's throttle to maintain a constant speed at all times. In the FM tuner, the d.c. voltage from the discriminator is fed back in the proper polarity to correct oscillator frequency and thus to retune the circuits so that the i.f. signal is in the center of the i.f. pass band. As in any governor system, there must be some deviation in order for any control voltage to be developed, so it is not possible to provide a 100 per cent correction. However, in a well-designed set the deviation can be held to less than 20 kc with ease. For the usual a.f.c. circuit, therefore, it is only the average d.c. voltage at the output of the discriminator which is employed for the control, and the average is obtained by filtering in the return circuit. As a matter of fact, we have seen sets in which the filtering in the a.f.c. return was inadequate, and some low-frequency audio signal was present in the control voltage. The net result was that when the a.f.c. was switched on the low-frequency audio voltage in the control signal would try to move the intermediate frequency back to the center again with each signal variation, and since the discriminator can only detect frequency variation, the low-frequency output of the set was reduced. So every time the listener switched the a.f.c. on, the bass dropped out of the reproduction.

Supposing, however, that the *audio* output from the discriminator were to be fed back to the reactance tube. If there were no filtering in the line, and enough control voltage were fed back, the output would be reduced appreciably throughout the entire audio spectrum. However, if a controlled amount of the audio signal were to be fed back, it would serve the same purpose as any other type of feedback and should therefore reduce distortion and improve frequency response.

This is the principle of the Dynamic Sideband Regulation as employed in the KN-120 tuner. Some of the audio voltage from the discriminator (actually that from a cathode follower which offers a low-impedance source for the voltage being fed back) as well as the filtered d.c. voltage (which is proportional to the average deviation from center frequency) is fed back to the reactance tube to give about 10 db of audio feedback throughout the i.f.

section and the discriminator. The result is that while the audio output is reduced by some 10 db, there is a considerable reduction in noise from weak stations, and the effect of over-modulation is reduced greatly. While FM stations by and large radiate a cleaner signal than is usually heard on AM, some of them do occasionally increase their modulation to improve coverage, and they may exceed the i.f. amplifier bandwidth in the tuner. There is a definite improvement in audio quality on most stations with the DSR turned on, and on fringe stations the signal-to-noise ratio is improved very noticeably. The FM section may be used without DSR, and with or without a.f.c., but since the operations are related, the DSR switch overrides the a.f.c. switch. Normal practice is to tune in a station with DSR and a.f.c. switches both at off; then switching the DSR switch to on also turns on the a.f.c.

General Description

Figure 9 shows the external appearance of the KN-120. Separate tuning knobs are used, and separate dial pointers and tuning indicators are used, as would be necessary on any AM/FM/Stereo tuner. In the lower panel, the lever switch at the left controls AM tuning, varying sensitivity and pass band for local, medium, or distant reception. At the right is another lever switch that selects the output fed to a pair of jacks marked OUTPUT on the rear apron—the FM jacks are normally connected to the FM circuit at all times. With this switch in the AM or STEREO positions, an AM signal is available at the OUTPUT jacks. One jack of each pair is fitted with a level-set control. The three slide switches on the front panel control power, a.f.c., and DSR.

Figure 10 shows the rear of the chassis removed from its case, which measures 4⅜ in. high, 15¾ in. wide, and 13⅛ in. deep. The case is steel, with a leather-like vinyl material permanently bonded to it. A multiplex jack is provided, and the built-in ferrite loopstick is mounted on a pivot so it may be oriented for optimum signal pickup.

In conclusion, it must be said that the DSR feature, along with over-all good design, results in a tuner which has adequate sensitivity and excellent audio quality. Furthermore, it is compact and attractive in appearance.

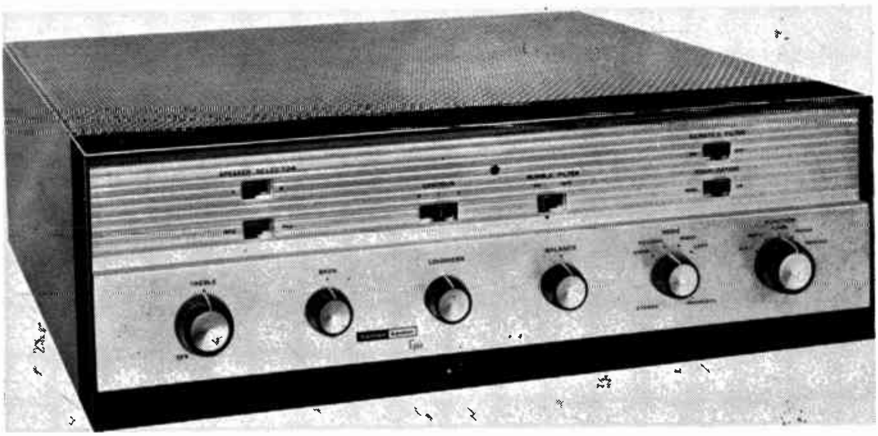


Fig. 1. The Harman-Kardon "Epic," dual 25-watt stereophonic amplifier system.

HARMAN-KARDON "EPIC" MODEL A250 STEREO AMPLIFIER

Combining compactness, simplicity of design and operation, and excellent performance into a single package capable of putting out a clean 50 watts total is somewhat of a feat, in our opinion. Many units accommodate two power amplifiers in a cabinet 15¼ in. wide by 13⅞ in. deep and 4⅞ in. high, while still others can encompass the multiple intricacies of a stereo preamp in slightly less than that space, but to combine both without crowding and with several very desirable features into a single unit rates considerable approval. The manufacturer describes the A250, shown in *Fig.* 1, as a *formidable* instrument, and while, to us, formidable implies one which would incite fear—or consternation regarding its use because of complexity—there is no reason for it. The amplifier is easy to handle and it certainly does provide most of the necessary functions.

Listing these functions, with reference to the panel arrangement, we note six slide switches along the top of the control panel—an extruded aluminum form, anodized and permanently copper colored. From left to right they are: two speaker selector switches, a three-position contour control, rumble filter, and at the extreme right the scratch filter and the equalization selector. The six knobs are, from left to right: TREBLE, BASS, LOUDNESS, BALANCE, MODE, and FUNCTION. The first four affect both channels simultaneously, while the two switches provide, for MODE: LEFT, RIGHT, MONAURAL, STEREO, and REVERSE; and for FUNCTION: TAPE HEAD, PHONO, TUNER, AUX 2, and AUX 1. Inside the unit and accessible from the top if cabinet mounted or from the rear if in its optional cage, is a SEPARATE-PARALLEL switch which ties the output amplifiers together and to the right preamp so the A250 may be used as a stereo preamp feeding a single built-in power amplifier (50 watts) for the right channel; the output of the left channel is available on a separate jack for feeding another basic amplifier. When this switch is in the PARALLEL position the two transformer secondaries must also be strapped in parallel.

The speaker switching is apparently unique to the Harman-Kardon line—we have noted it before in the A224 "Trio." Each amplifier terminates in an output transformer with secondary impedances of 4, 8, 16, and 32 ohms, one end of the winding being grounded. Two additional terminals, marked A and B, are provided for each amplifier. These are connected to one of the speaker selector switches, similarly marked A and B, which grounds either the A or B terminals at the user's choice. A and B are used for the ground returns of two separate pairs of speakers, located in different rooms, perhaps. Thus either pair can be energized at will. Also, if desired, the second speaker selector switch may be set to ALL, instead of ONE, both sets of speakers will operate at once. The reason for the impedance range extending to 32 ohms is that when both output sections are paralleled, the speaker is connected to the tap twice its nominal impedance, which necessitates 32 ohms for a 16-ohm speaker, and so on.

Both channels are identical, and employ 12AX7's as phono/tape-head preamplifiers, 12AU7's as tone-control amplifiers, 12AT7's as amplifiers and phase splitters, and two 6L6GB's in the output stages, the latter being mounted at an angle of about 40 deg. in order to maintain a low silhouette in the cabinet. Tone controls are of the Baxendall type, which we consider most desirable, and phono and tape-head equalization is derived from feedback over the two stages of the preamp, and accommodates RIAA and EUR on phono, 7½ and 3¾ on tape. A 47,000-ohm load is provided as a fixed value for the phono cartridge or tape head, while a 100,000-ohm load is provided for ceramic cartridges, followed by a 26-db voltage divider. The scratch filter operates only on phono, while the rumble filter operates on all inputs—which we believe is desirable because some radio stations require rumble filtering for best listening results. The two contour curves turn over at about 350 and 700 cps, respectively.

Plate supply is furnished from a voltage doubler circuit using the new silicon rectifiers, and resulting in extremely low hum levels. Plate currents in the output stages may be balanced with the controls provided, thus further lowering hum, and with d.c. on the first three stages the hum on phono is better than 62 db below a 1-watt output.

The power supply fuse and two a.c. receptacles are mounted on the rear apron, together with the two output terminal strips. All inputs are located on the top of the chassis, side by side in two separate rows. Four shorting plugs are furnished for insertion in unused inputs, and a plastic clamp is located on the rear of the chassis adjacent to the input jacks so that all leads to the amplifier may be dressed neatly where they come out.

Performance

The amplifier has more than adequate gain. For a 1-watt output, and gain control at maximum, signals of 0.9 mv are required at the magnetic phono input and 20 mv at ceramic phono input, 0.35 mv at the tape-head input, and 52 mv at auxiliary and tuner inputs. At a 117-volt line, distortion was 0.85 per cent at 25 watts on one channel, 0.87 per cent on the other. Control tracking varied from a maximum of +3 to −2 db on volume, and from +4 to −3 db on tone at six points checked on both. At the specified input signals, the TAPE OUT jack provided a 0.45-volt signal to feed a tape recorder, unaffected by tone or volume controls.

On listening tests, the A250 confirmed the measurements as to sound quality, and as has often been said, "measurements mean little if the listening is bad." Using several different speakers and a widely varied range of program material, the amplifier performed admirably and even after hours of use it was still necessary to take the eggs into the kitchen to fry them—all three transformers run cool. Actually, this was to be expected in the output transformers at least, since the cross section of the cores measure 1 11/16 square, which is plenty for a 25-watt transformer. On the whole this is a unit of excellent performance and appearance, and should result in an equally excellent over-all system.

Fig. 1. The Harman-Kardon "Trio," Model A-224—a multipurpose stereo-monophonic-conversion amplifier.

Harman-Kardon "Trio" Tri-plex Stereophonic Amplifier

CONVERSION TO STEREO has become the principal problem to many audiofans —mainly because they are not yet completely familiar with what facilities they are likely to require. Either they have a complete monophonic system already and feel that from an economic standpoint they must use as much of it as possible, or else they plan to start from scratch, adding from time to time as they become more familiar with their needs.

The Harman-Kardon "Trio," Model A-224, serves—as its name indicates—in three roles. First, it is a complete stereo amplifier system with 12 watts of audio in each channel; second, it is a monophonic amplifier with 24 watts of audio (leaving the preamplifiers and tone-control stages unused); and third, it is a conversion amplifier which can serve as the input section of a stereophonic system and one of the two output amplifiers—one which provides 24 watts of power—with the other being one already in the user's monophonic system.

In its first role—that of a complete stereophonic amplifier—the Trio provides two equalized preamplifier stages, usable for either phono or tape head inputs, and also accommodates three high-level inputs, such as tuner, tape preamp output, or high-output phono pickup, such as a ceramic. In the latter inputs, the impedance offered to the cartridge is 2 megohms, which is sufficient for good low-frequency response with a ceramic cartridge. The preamplifiers are 12AX7's, with feedback equalization, switchable on the rear panel of the amplifier to phono or tape positions. These are followed by one section of a 12AU7 in each channel to provide sufficient gain for the tone controls, and these are followed in turn by 12AX7's as gain and phase-splitter stages, which drive a pair of EL84's in each output channel.

Tone controls are ganged for the two channels, with bass and treble being separate as in all high-fidelity equipment. The volume-loudness control is made to serve in the chosen role by means of a slide switch, and another slide switch serves to control the rumble filter. The MODE switch provides for stereo, stereo reverse, and either right or left input channel to both outputs as a monophonic amplifier. The FUNCTION control selects the input source.

Two other slide switches provide an unusual form of control which would be ideal in many installations. One switch, with positions labeled ONE and ALL, controls whether one or two pairs of output lines are being fed, and the other controls which of the two output pairs is being fed when the first switch is set at ONE. Thus the user can feed one speaker system in his living room or another in his den, or he may feed both at the same time if he wishes. This provides a sort of flexibility in speaker control that is rarely found on an amplifier control panel. Similar control action works as well when the amplifier is used in the monophonic application.

For both monophonic and stereo-conversion applications, it is necessary to switch the operation from SEPARATE to PARALLEL. This is done by means of a slide switch located inside the amplifier on the apron behind the front panel. This switch connects both output amplifiers to the right input section, and feeds the left input section to an output jack located on the rear panel of the amplifier. When this is done, it is necessary to strap the speaker connections of both output sections together. Both amplifiers have output impedances of 8, 16, and 32 ohms, and the instructions indicate that when a 16-ohm speaker is used in the monophonic or stereo-conversion application, both 32-ohm terminals should be strapped together to feed a 16 ohm speaker and so on. In the usual manner, the second (left) channel is fed to the input of a second power amplifier and thence to its own speaker system. The speaker selector switches are inoperative in the stereo-conversion connection, but they still function in the monophonic use.

Sensitivity of the amplifier is such that a 3-mv signal at the phono input or a 1-mv signal at the tape input will give the rated 12 watt output, and a similar output is had from a 300-mv signal at the AUX, TUNER, and high-level phono inputs. A tape output is provided ahead of both tone and volume controls, with a 1-volt signal for the indicated inputs. The output for the left-channel external amplifier is 0.5 volts. Tone control ranges are $\pm$12 db at 50 and 10,000 cps, with the rumble filter giving a rolloff of 12 db per octave below 50 cps.

The Trio may be used in its normal metal housing (at extra cost) or it may be mounted behind a conventional panel without the enclosure. In operation, it was found to be convenient, and to have adequate control flexibility for practically any installation. The only change we might suggest in the entire unit is that it might provide both tape and phono input jacks for the preamplifiers, and to have the FUNCTION control switch inputs from tape to phono (and change the equalization at the same time) as a front panel control, rather than making it necessary to change input plugs and throw the switch on the rear panel. This is an amplifier we could recommend heartily for any one going through the pangs of conversion from mono to stereo.

THE FISHER MODEL 400-C STEREOPHONIC AUDIO CONTROL

There has never been any question in the minds of well informed audiofans as to the generally high quality of Fisher equipment, and the 400-C Master Audio Control continues in the same tradition. This is an attractive self-powered unit requiring a cutout 4½ in. high by 14⅞ in. wide and extending back from the panel some 7½ in. It is available as a chassis for mounting in an existing cabinet as well as in wooden cabinets in various finishes, one being shown in *Fig*. 3.

This preamp-control unit has a number of unique characteristics, in addition to all of the normal facilities expected in a stereo control unit. One of the most interesting from the standpoint of the user who may have both changer and turntable is the equalization selector. Two phono inputs are provided—and the requirements have been considered thoroughly by the designers to give the utmost in simplicity of operation. One of the inputs has three degrees of equalization—RIAA, EUR, and LP, with the latter two functioning only for monophonic pickups. The RIAA position on this input may be used for either mono or stereo records. The second phono input has RIAA equalization only, and connections to two separate pickups may be left in place permanently even though the user has two record playing units.

The TAPE input is for connecting directly to the tape heads, and correct equalization is built in so that all controls will be centered in the 400-C. The MIC input is flat, and arranged to accommodate any high-impedance microphones. Both TAPE and MIC inputs may be used for stereo or mono, as desired.

Provision is made for three high-level inputs, all dual-channel, and selection between inputs is made by a push-button

Fig. 3. The Fisher Model 400-C stereophonic Master Audio Control.

switch, with pilot lights indicating which input is in use. A rumble filter is provided, with low-frequency cutoffs at 20, 50, and 100 cps, and the loudness contour is selected by another switch which gives two degrees of compensation as well as a flat position.

The remaining controls are: channel balance, bass, treble, volume and power switch, and an output selector switch. The latter has six positions: normal and reverse stereo, channel A feeding both outputs, channel A feeding channel A output alone, channel B feeding both outputs, and crossover. This last is just what the name implies, with a 650-cps crossover network switched in so that the channel A output has the highs rolled off and the Channel B output has the lows rolled off, both being fed from the channel A inputs. This permits a stereophonic effect from mono sources, since the two speakers are reproducing different frequency ranges with a consequent distribution of the sound between the two channels.

The tone controls are the Baxendall type —which we prefer—and the contouring of loudness compensation is pleasant. Four a.c. output receptacles are provided, all being controlled by the power switch, and accommodating a total of 650 watts which should be adequate for two amplifiers and a turntable and/or changer.

All heaters (except that of the rectifier tube) are fed from d.c. which is provided by a selenium rectifier and a filter system consisting of two 1000-μf capacitors and a 10-ohm resistor. The plate-supply filter is a two-section RC type, and residual hum is better than 80 db below normal output of 2 volts for high-level inputs, and better than 60 db below the normal output for the low-level inputs. Tubes employed are: four ECC83/12AX7's, two ECC81/12AU7's, and one EZ80/6V4. The front panel is brushed brass, and is almost ⅛ in. thick. Cathode-follower outputs are provided for a tape recorder, feeding from both channels, and tape monitor jacks are furnished for use with recorders having feed-through connections built in.

Construction

Figure 4 shows the top and rear of the chassis for the locations of input and output jacks. The Channel A connections are on the top row and Channel B connections are on the lower row, except for the tape monitor jacks which are shown at the left of the top row of jacks. Level-set controls are used for the low-level inputs through the preamplifier, and for the AUX 1 high-level inputs. These are shown on the housing in front of the tubes, and are accessible from the top. Both preamplifier tubes, shown on the platform at the right, are shock mounted for complete freedom from microphonics.

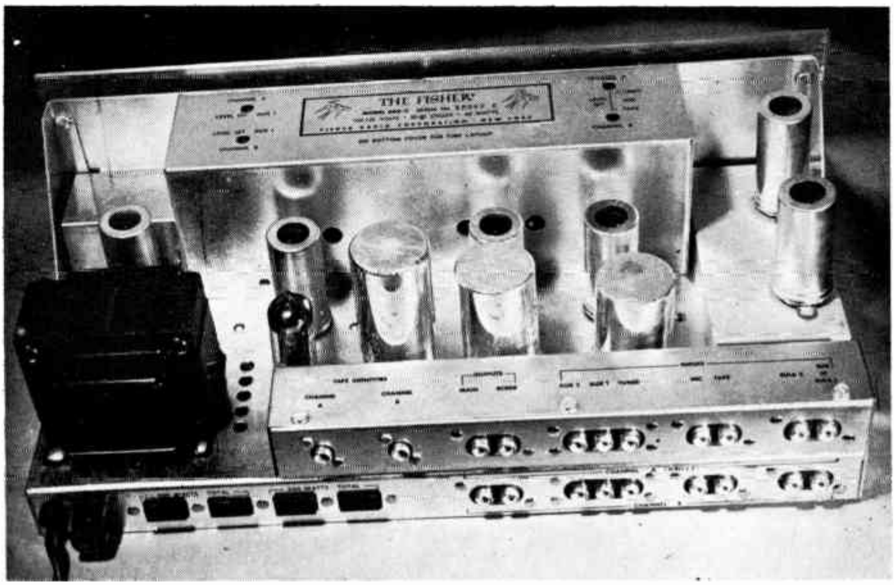


Fig. 4. Chassis view of the Fisher 400-C shows input jacks, level-set controls, and general layout.

The 400-C is a real joy to use. The switching is quiet, with no noise between switch positions; controls are quiet and tapers have been chosen for smooth operation. The controls have a firmness of feeling, which gives the same impression as when handling a fine car. Distortion is low—measuring around 0.1 per cent at a 1-volt output, and the equalizations of the two preamplifier sections in the model tested was closer than 1 db of being identical in both channels. All in all, we consider the 400-C to be one of the finest examples of preamplifiers we have been privileged to examine.

THE BELL "CARILLON" STEREO AMPLIFIER

SEVERAL YEARS AGO, long before announcement of the stereo disc heralded a new era in audio reproduction, Bell was one of the very first—perhaps *the* first—to market an integrated stereo amplifier. The Carillon is the latest succession to Bell's original model and contains a number of refinements of features and design, provides about three times as much power, is good to listen to, just as good to look at (as handsome a unit as this reviewer has seen), and, considering its power, is relatively light and compact.

The Carillon provides almost all the special stereo features and functions that can prove useful. There is a balance control, a master gain control, and a function switch with three positions: stereo, reverse stereo, and monophonic. The first two positions are self-explanatory. In the mono mode, the two channels are combined, which is desirable when playing a mono disc with a stereo pickup, for this causes the audio signals (lateral information) to add and rumble (vertical) to cancel. If it is desired to feed one source to both channels, the function switch is set to mono and the balance control is turned to one extreme or the other, depending upon whether the left or right source is to be cut off; since the balance control permits infinite attenuation, this results in shutting off one source.

To balance the left channel against the right all the way from the program source (disc, tape, or tuner) to the sound emanating from each speaker, it is highly desirable to be able to alternate rapidly between two conditions: left channel on and right channel off; left channel off and right channel on. The Carillon allows one to do exactly this, although the instructions fail to mention the fact. There is a speaker switch that connects speaker A or speaker B or both; this applies to each channel.

The original purpose is to enable one to connect an additional speaker, to each channel for the den, playroom, or wherever. On the output terminal strip for each channel, there are terminals marked A and B, intended for the speaker in each room. If instead one connects the left stereo speaker to terminal A on one terminal strip, and the right stereo speaker to terminal B on the other terminal strip, switching between positions A and B will alternate the sounds of the two speakers. In the A-B position of the speaker switch, both speakers will be on. (If one employs the switch for balancing at high levels, there is the possibility of damaging the output transformers if their loads are suddenly removed. Therefore it is advisable to connect a 50-ohm 10-watt resistor across each output transformer. Power consumption by this resistor will be relatively slight.)

The only significant omission in terms of stereo functions is that of phase reversal. Fortunately, this omission is easy to correct by installing a double-pole double-throw switch at the speaker or elsewhere to reverse leads to one of the speakers, although for stereo records this function no longer seems to be necessary.

For each channel, there are seven outputs. Three are high level, and two of these—TUNER and TAPE AMPLIFIER—have input level sets. The low-level inputs are for high-impedance microphonic (magnetic), tape head, magnetic phono cartridge, and ceramic phono cartridge. Although ceramic pickups are actually high-level, amplitude devices, in the Carillon the signal from such a cartridge is fed through a small capacitance, which in effect converts the cartridge into a low-level, velocity device, so that the same preamplification and equalization may be employed as for a magnetic cartridge. The amplifier can accommodate either a ceramic or magnetic pickup, but not both, inasmuch as the input jack for each leads to the same point on the selector switch.

There are no level sets for the low-level inputs. To this reviewer's way of thinking, a level set following the preamplifier stage would be useful for equating the volume obtained from a low-level source with that from the high-level sources and for achieving balance between the two sections of a stereo phono pickup (differences between sections can be as much as 4 to 6 db) or the two sections of a stereo tape head. Inasmuch as most tape amplifiers and most tuners contain gain controls, which in a pinch can serve the same purpose as input level sets on the amplifier, it would seem preferable if one of the two pairs of level sets in the Carillon had been used for the preamplifier section instead.

The Carillon follows the trend toward separate rather than ganged tone controls, permitting one to compensate for differing tonal characteristics of unlike speaker systems, for different effects of room acoustics upon each speaker, for different tonal balance in each channel of the program material, and so on. On the other hand, the low filter for reducing rumble and the high filter for reducing noise are ganged devices, each one controlling both channels.

The Carillon is rated at 30 watts per channel. This reviewer measured about 25 watts output at mid-frequencies before clipping became apparent on an oscilloscope. The difference between 30 and 25

Fig. 1. Bell "Carillon"—a new stereo amplifier.

watts is quite minor, less than 1 db. At the frequency extremes of 20 and 20,000 cycles, each channel was able to deliver 20 watts of a well-formed sine wave before clipping occurred, and this is very good. Clipping was symmetrical, and the amplifier gave no sign of distress, such as oscillation or radical change in waveform, when driven into the clipping region—also very good.

Circuitry of the Carillon is straightforward and follows design principles of proven worth. The unit may be termed an "all-feedback" amplifier inasmuch as there is feedback in the preamplifier (between two sections of an ECC83/12AX7) for equalization, feedback in the following stages (ECC83) in connection with tone controls and the high filter, and feedback from the output transformer over the remaining stages.

The left channel provides equalization for the LP and "European" phono recording characteristics as well as for the RIAA curve. The right channel supplies only RIAA compensation. The thought is that all stereo discs are recorded RIAA, so that LP and European equalization are needed only for mono discs, i.e., only for one channel. But, as previously pointed out, the best way to play mono discs with a stereo cartridge is to parallel the outputs of both cartridge sections in order to cancel vertical rumble. However, if one sets the Carillon's selector switch to the LP or European position, equalization will instead be RIAA on the right channel, with a consequent tonal unbalance between channels. This would be alleviated if the user switched cartridges to a mono pickup (properly wired to the head) when playing mono records.

The high-level sources and the output of the preamplifier stage feed into the selector switch. Connected to the arm of the switch are: an output jack for feeding a tape recorder; the low filter circuit; and. the high end of the volume control. The signal goes from the arm of the volume control to the balance control, to the function switch, and to the first grid of the ECC83 associated with the tone controls and high filter. Following this tube is the loudness control and then the power amplifier section, using an ECC83 as a voltage amplifier and split-load phase inverter. The output stage employs 6CA7's connected in Ultra Linear fashion and using cathode bias. This results in less power output than with fixed bias (some amplifiers obtain 50 watts and more with 6CA7's), but on the other hand one does not run the same risk of tube destruction if the bias supply should fail.

All heaters but those of the output tubes are d.c. operated. This is done by connecting the cathodes of all four output tubes (for the two channels) in parallel and running part of the cathode current through the heaters, which are arranged in a series-parallel configuration.

In a stereo amplifier, it is important not only that each channel provide good performance in terms of frequency response, distortion, equalization, etc., but also that the two channels be very similar to each other in performance characteristics. Following are measurements taken by this reviewer with respect to a number of functions, showing that on the whole the Carillon maintains excellent correspondence between channels, along with very creditable performance in absolute.

1. *Treble controls.* At 15,000 cps, the left channel provided a maximum of 9 db boost and a maximum of 21 db cut. For the right channel, the respective figures were 10 db and 20 db.

2. *Bass controls.* At 30 cps the left channel provided a maximum of 16 db boost and a maximum of 18 db cut; the right channel, 15 db and 19 db.

3. *Master loudness control.* Following were the amounts of bass boost at 50 cps relative to 1000 cps for various degrees of attenuation:

Attenuation At 1000 cps	BOOST AT 50 CPS Left Channel	Right Channel
10 db	6 db	6.5 db
20	12	13
30	18.5	19
37.5	24	25

Unlike most other loudness controls, which provide unlimited attenuation, the Carillon's loudness control confines attenuation to a rated 40 db (actually 37.5 db according to the reviewer's measurements). This should be sufficient for virtually all circumstances. Limited attenuation makes it possible for the control to maintain the excellent balance indicated by the above figures. The importance of such balance cannot be overemphasized.

4. *Master gain control.* Its ability to maintain balance between channels was very good on the whole, but, because it permits infinite attenuation, not quite so good as that of the master loudness control. During the first 5 db of attenuation, balance in the reviewer's unit changed by 3.5 db, with the right channel dropping in level below the left one. But between 5 db and 55 db attenuation—which is apt to be the working range 99 per cent of the time —balance remained virtually within 2 db. Thus if one were to adjust the balance control for equality between channels with the master gain control at mid-setting, one would have inter-channel balance within 2

db at any normal setting of the gain control. It may be pointed out that even at 60 db attenuation, one would have balance within 3.5 db.

5. *Low filter.* This provided the following attenuation at selected frequencies:

	Frequency	ATTENUATION Left Channel	Right Channel
40-cps position:	100 cps	.5 db	.5 db
	40	4	3
	20	9	7.5
80-cps position:	200	.5	.7
	100	2	2
	40	7	8
	20	13	15

6. *High filter.* This provided the following attenuation at selected frequencies:

	Frequency	ATTENUATION Left Channel	Right Channel
8000-cps position:	4,000 cps	1.5 db	2 db
	8,000	10	10
	15,000	15.5	15
4000-cps position:	2,000	1	0.5
	4,000	8.5	7.5
	8,000	16.5	15
	15,000	21	19

7. *RIAA equalization.* The difference between channels was 2.5 db at 50 cps and only 0.5 db at 10,000 cps. The maximum absolute error was 3 db excess bass boost at 50 cps and 2 db insufficient treble cut at 10,000 cps.

8. *NARTB (tape) equalization.* The difference between channels was 2 db at 50 cps and only 0.5 db at 15,000 cps. The maximum absolute error was 3.5 db insufficient bass boost at 50 cps and 2 db insufficient treble cut at 15,000 cps. Since most playback heads have at least 1 or 2 db loss at 15,000 cps, the variation at the treble end is of no consequence.

9. *IM distortion.* The following readings were obtained at various amounts of equivalent sine wave power, using 60 and 5000 cps in 4:1 ratio:

Equivalent Sine Wave Output	IM DISTORTION Left Channel	Right Channel
1 watt	0.4 %	0.25%
5	0.5	0.65
10	0.65	0.8
15	0.8	0.95
20	1.05	1.2
25	1.8	1.6
30	3.6	2.55

10. *Sensitivity.* As measured on the basis of a signal fed into the tuner input and with the master gain control full on, sensitivity on the right channel was about 1.5 db lower than on the left channel.

11. *Frequency Response.* With the tone controls set to mid-position and with the master gain control at maximum, frequency response of each channel was extremely flat, better than within 1 db, from 20 to 15,000 cps. With the gain control set for 6 db reduction in volume, the worst position for high-frequency response, there was a drop of 1.5 db at 10,000 cps.

Based on 25 watts output at 1000 cps, the reviewer measured a signal to noise ratio of 74 db on high-level input, which is superior to the manufacturer's claim of 71 db. The ratio measured 55 db on magnetic phono input in RIAA position, 48 db on tape head input, and 61 db on microphone input. Crosstalk between channels measured 66 db isolation at 1000 cps, 61 db at 50 cps, and 45 db at 10,000 cps.

As a final measurement, the reviewer fed square waves into the amplifier and viewed the output on an oscilloscope for signs of ringing. None at all were evident at 100 and 1000 cps, and only one slight ripple was observed on a 10,000-cps square wave.

Considering the complexity of a stereo amplifier, a well-written instruction book is of considerable importance to the purchaser's successful use of the unit. The Carillon comes with such a book, which contains four drawings showing how to connect one set of stereo speakers, how to connect two sets of stereo speakers, how to wire a complete stereo system to the Carillon, and how to use the Carillon as a monophonic 60-watt amplifier. Even though the booklet is written on a completely nontechnical level, it contains a schematic, a laudable recognition of a fact of life—that any piece of audio equipment may some day require servicing and that the serviceman to whom the equipment is brought may not have the schematic on hand.

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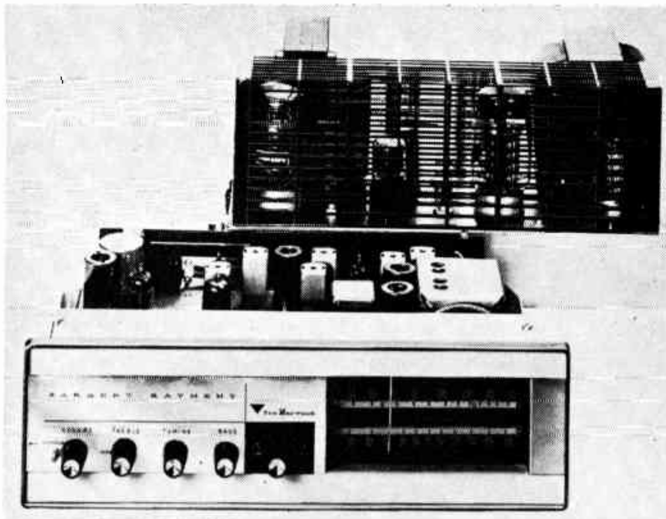
Fig. 3. Sargent-Rayment 300-M70 tuner and amplifier combination.



SARGENT-RAYMENT 300-M70 TUNER-AMPLIFIER COMBINATION

As may be recalled from years back, we have long been enthusiastic about the Sargent-Rayment line of tuners—their earlier SR-58 being far and away the best AM-only tuner we have ever heard with the exception of the Miller Wide-Range circuit which was available only as a kit some years ago. The SR-68, an AM-FM tuner was practically the Cadillac of tuners.

In keeping with the trend toward smaller and more compact models—and also toward table-top housings, S-R has recently introduced a series of tuners and amplifiers of unusual physical design which are well suited to current decorating schemes. As seen in *Fig.* 3 the tuner is shown with a conventional front panel to permit installation in the usual type of cabinet. However, when the user wishes to have the tuner in a table-top cabinet, he simply purchases the cabinet and the conversion panel, removes the console panel and ends up with the sloping front "cantilevered" cabinet. This arrangement is attractive, and provides two possibilities for use.

There are three series of tuners—the SR-100, which is just a tuner, but equipped with a volume control and a bandwidth selector to give high-fidelity AM reception; the SR-200, which is combined in one cabinet with a 25-watt amplifier; and the SR-300-M70, which has the tuner with preamp and tone controls in one unit, and a 70-watt power amplier in the second section. Power for the tuner comes from the amplifier unit, eliminating high a.c. fields and the heat-producing rectifiers and output tubes from the tuner chassis.

The tuner itself is attractive in appearance and performs well. Sensitivity is rated at 3 μv for 20 db quieting, and the manufacturer gives distortion figures, which in itself is unusual, that indicate excellent performance. Listening quality is outstandingly good on FM, and about normal on AM. Provision is made for FM, crystal, and magnetic cartridges, with three equalization positions available—AES, LP, and RIAA.

The 70-watt amplifier employs the British-made KT-88's in an Ultra-Linear output stage with fixed bias and a balancing control. Tube currents can be metered separately, and output impedances of 4, 8, and 16 ohms are available. The first stage is an EF-86, and it is directly coupled to the input grid of a 6SN7 in the familiar ''long-tailed pair'' phase splitter circuit—a modification of the circuit generally referred to as the Mullard, but basically the same as that used in the Leak TL/12 amplifier, which is some eight years old, at least. Plate power is furnished by two GZ-34's, and resistance-capacitance filtering is used for all the lower-voltage stages. Hum level was measured at 58 db below 1 watt, which is slightly better than the 90 db below rated output which is claimed by the manufacturer.

The tuner connects to the power amplifier through a 4-foot cable which is equipped with octal plugs and receptacles, and additional lengths are available for installations where the tuner is required to be further from the amplifier than the original cable allows.

The tuner tone controls are effective, and we particularly like the loudness control, which is a small handle rotating around the volume-control knob. We are still firmly of the belief that some loudness contouring is essential to any good home music system today, and to be effective it must be easy to handle and must be contoured to match the ear's response. However, there are many people who do not share this opinion—at least the fact that both flat and contoured controls are available give the listener his choice.

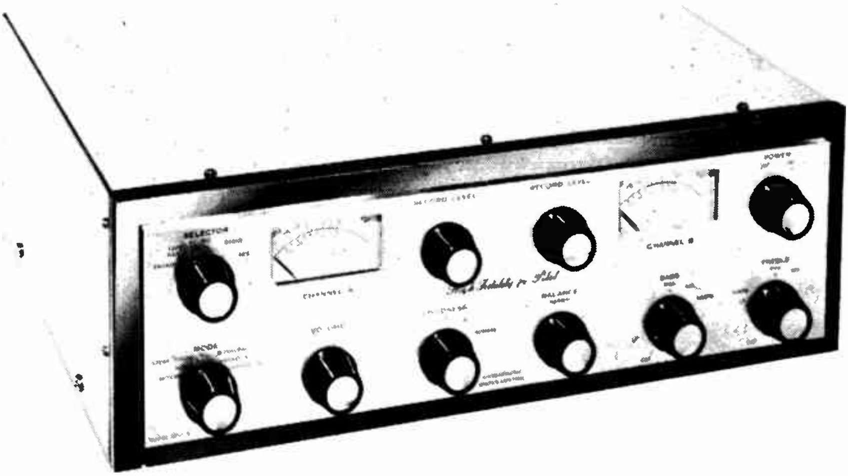


Fig. 7. Pilot SP-215 stereo preamplifier is a top-quality unit flexible enough for normal preamp use as well as for various recording applications.

PILOT STEREO AMPLIFIERS—SP-215 AND SM-244

One of the first of the stereo amplifiers to appear on the market was the Pilot SM-244, and along with it came the preamplifier, SP-215. Both offer good performance, and the preamp may be used as the nucleus of a complete high-fidelity record and playback system.

The SP-215, shown in *Fig.* 7, is extremely flexible, well engineered, and carefully built. Briefly, the circuit consists of two separate sections, both alike, and serving as the two channels of a stereo system. As to the circuit, there are five inputs to each channel—phono, tape head, microphone, radio, and auxiliary. The first three are preamplified, with suitable equalization for phono (RIAA) and tape head (NARTB). The selector switch connects the desired input to the following circuitry, which is split into the recording section and the audio section. The former has a separate gain control feeding an amplifier stage, which in turn feeds a cathode follower for the recording output and a VU meter amplifier stage. The audio section begins with one section of a dual volume control, (a four-section potentiometer with two sections in each channel), followed by a loudness contour switch with five positions, a balance control, voltage amplifier stage, tone controls, another voltage ampilfier stage, the second section of the dual volume control, and a cathode follower for audio output.

In addition, there is a function switch which provides for normal stereo operation, reversed stereo in which Channel A input is fed to Channel B output and *vice versa,* and two monaural positions in which both

Fig. 8. Pilot SM-244 offers both preamplifier and power output for stereo use.

outputs are fed from either Channel A or Channel B inputs. Furthermore, a socket is provided for powering an external record amplifier which would provide the necessary equalization for recording with the specific heads employed.

With the audio circuits being ganged together, stereo operation is permitted with a minimum of controls. And with completely separate controls of the two recording outputs, flexibility of operation is assured in this circuit as well.

Circuit engineering is to be commended in this unit, with all heaters in the audio channels operating from a d.c. supply. The recording channels—which operate at a higher level—employ a.c. on the filaments. The ganged controls have been held to close tolerances, with variation in gain not exceeding 1 db between channels.

Gain in the preamp stages is altered so that in the phono position a 7.5 millivolt signal is required for normal output, while only 2.5 mv is required for normal output on tape-head and microphone inputs. Separate level adjusting controls are provided for the two high-level inputs of each channel, and another dual control permits setting the record output signal to a value which will give an adequate indication on the VU meters.

Distortion is less than 0.25 per cent for a 1-volt output, and hum and noise measured 78 db below 1 volt. The normal output for the audio channel is 1 volt, while the recording output may be set anywhere from 0 to 1.3 volts. The tone controls—which affect the audio channel only—are marked with calibration points which correct for LP, NAB, and AES curves without complicating the input switching for the lesser used equalizations.

This is a unit which will delight the eye of anyone who admires a good instrument, and its many uses should make it an extremely popular amplifier.

The SM-244 Amplifier

Similar in appearance, except for the VU meters and the recording controls, the SM-244, shown in *Fig.* 8, combines the preamplifier equipment, tone controls, and the balance and contour controls of the SP-215 with two 14-watt power amplifiers to serve as a complete stereo system. The circuits are identical except for the output circuitry —the SM-244 has no meters, but does provide cathode follower outputs for both channels to feed a recorder, and the audio sections terminate in the power amplifier consisting of a 12AX7 and two EL84's in each channel, with output impedances of 4, 8, and 16 ohms. There would never be any need for the use of an SM-244 with an SP-215, since the latter could feed two basic amplifiers to provide a complete system. But for anyone who did not require the flexibility of the recording channels, the SM-244 would serve admirably.

Distortion—IM—measured 2 per cent at 16 watts in one channel and at 17 in the other in the model tested. Hum and noise was measured at 66 db below 1 volt, which is quite satisfactory. While we prefer the SP-215 with two higher-power amplifiers for a complete system, the SM-244 should be perfectly adequate for any average installation where only a single pair of speakers was to be used.

MADISON FIELDING SERIES 330 STEREOTUNER

With the growing number of FM-AM stereocasts, demand has increased for stereotuners—separate AM and FM tuners on a single chassis. Today in approximately two dozen cities there are affiliated FM and AM stations that are stereocasting on schedules ranging from one-half to 20 hours a week. One of the newcomers to the still slender ranks of stereotuners is the Madison Fielding Series 330 shown in *Fig.* 1. This unit has four output jacks, which supply the following signals:

1. FM, for feeding the FM signal into the left channel of a stereo amplifier.
2. AM, for feeding the AM signal to the right channel of a stereo amplifier.
3. AM/FM, for feeding either the AM signal or the FM signal into a single-channel amplifier.
4. Multiplex, which takes the FM audio signal prior to the treble de-emphasis circuit for the purpose of supplying a multiplex adapter, which will be on the market when multiplex broadcasting becomes a reality.

The Series 330 is neat and simple in appearance, with only three operating knobs, one for AM tuning, one for FM tuning, and the third a selector switch with four operating positions, as follows:

1. OFF.
2. AM: The AM signal is connected both to the AM/FM output jack and to the left channel output jack, marked FM. Moreover, although the instructions make no mention of this, the FM signal is connected to the right channel output jack, marked AM. In others words, in the AM position of the selector switch the channels are reversed.
3. STEREO: Here the FM signal is connected to the FM output jack and the AM signal to the AM output jack.
4. FM: The FM signal is fed to the left output jack, marked FM and to the AM/FM output jack. There is no signal present then at the AM output jack.

It would seem that the above variations as to the signals that may be obtained at the various output jacks would take care of most needs of most stereofans. Moreover, there are level controls for the two output jacks at the rear of the chassis, permitting output levels to be brought at least approximately into balance. Exact balance cannot be achieved, except for two specific FM and AM stations because levels differ among stations and, in the case of AM, with signal strength.

FM sound is clean and, when A-B'd with a comparison tuner of known characteristics, seems to have the correct amount of treble de-emphasis. An appreciable number of manufacturers of FM tuners succumb to the temptation of using less than the required amount of de-emphasis, thereby imparting a false brilliance to the sound, a brilliance which often does not wear very well upon protracted listening.

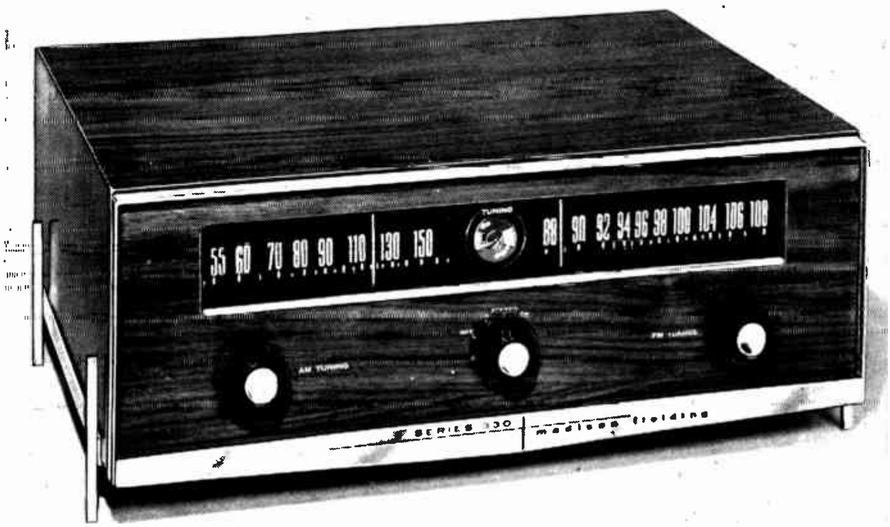


Fig. 1. Madison-Fielding Series 330 Stereotuner.

The FM circuit is on the whole simple and conventional, with a few departures from convention that make a favorable difference. It employs a tuned grounded-grid r.f. stage, a triode mixer and separate triode oscillator, a triode a.f.c. circuit, two i.f. stages, a limiter, and a ratio detector, which is no longer looked upon with disdain for use in first-rate tuners. One of the things that sets the Series 330 apart is the unusually wide i.f. bandwidth, which is 355 kc at the 3-db-down points. This helps keep distortion low and accounts at least in part for the clean character of the sound. Alignment can be performed without removing the bottom plate, so that this does not raise the problem of alignment changing when the bottom plate is put back. Sensitivity is high, and a 3-foot strand of wire appeared to work quite satisfactorily at a distance of 30 miles from a number of FM stations. A.f.c. action is quite moderate, just enough to overcome drift but not so much as to complicate seriously the problem of selecting a weak station adjacent to a strong one. Use of a tuning eye—half of a 6AF6 (the other half is used for the AM section)—facilitates tuning, and the point where a station comes in best coincides with maximum closure of the eye.

All in all, the FM tuner appears to be a very satisfactory unit.

While in the main the same can be said of the AM tuner, the latter does raise one serious question, that of adequate treble response. Many AM stations transmit a wide range signal, extending in a number of cases to 10,000 cps, 12,000 cps, or even higher. The limitation on frequency response, therefore, often lies in the AM tuner. The Series 330 appears to have what is often called "typical AM sound" so far as frequency response is concerned. This reviewer tuned the AM section of the Series 330 to a New York City AM station known to transmit a wide-range signal, and there was a very decided difference between the signal received on the AM section of the tuner and the signal received on the FM section, which was tuned to the FM adjunct of the station.

Otherwise, there appears little if anything to criticize. Within the limitations of frequency response, sound is clean, attributable in part to the use of a separate diode for a.v.c., thereby minimizing distortion. Also, the use of a separate diode makes the a.v.c. action more effective, sc that the volume level is more constant from one AM station to another.

Cathode-follower outputs are used for both the FM and AM sections, permitting long runs of cable to the stereo amplifier. Flywheel tuning facilitates station selection. The unit has a low silhouette and, as previously remarked, is good looking; it becomes even more handsome when installed in the natural wood finish cabinet available at extra cost.

MADISON FIELDING SERIES 320 STEREO AMPLIFIER

The sudden onrush of stereo due to the emergence of the stereo disc doubtless has caused amplifier manufacturers to do a good deal of deep thinking. In designing an amplifier for stereo, there is much room for the display of imagination, originality, and ingenuity, for there are many ways in which the amplifier can provide for controlling the stereo channels, coordinating them, and permitting their use for monaural as well as stereo sources. The Madison Fielding Series 320 amplifier reflects a good deal of serious and imaginative thinking as to the problems that the audiofan—or rather stereofan—is apt to encounter.

The Series 320 includes a control amplifier (tone, gain, loudness, selection) and a power amplifier, rated at 20 watts, for each channel.

The 320 is as much interested in providing monaural[1] service as stereo service—after all, most of us will still want to keep playing our treasured LP's and to keep listening to single-channel FM until multiplex comes along. To this end, the selector switch has duplicate sets of positions, one set to the left of the center position, and the other to the right. When turned to the left (where there are three positions marked tape, tuner, and preamp), the amplifier becomes a monophonic device, causing any input fed to the left channel also to be fed to the right channel. Inputs to the right channel are then disconnected. When the selector switch is turned to any position to the right of center (again there are three positions marked tape, tuner, and preamp), then the inputs to each channel are normalled through; that is, inputs to the left channel go through the left channel amplifier, and inputs to the right channel go through the right channel amplifier. In short, when the selector switch is right of center, the Series 320 is a stereo amplifier.

Unlike the majority of stereo amplifiers—although there are other exceptions in the same respect—the Madison Fielding Series 320 does not have a balance control. Instead it relies upon separate gain controls for each channel to achieve balance. Concentric with each gain control is a switch that converts the former into a loud-

[1] The term monaural is used here instead of monophonic because the former is the term used on the panel of the Series 320.

ness control. Like all stereo amplifiers (at least all this reviewer has come across), the Series 320 has a master gain control that governs both channels simultaneously.

One of the most intriguing and novel features of this amplifier is the means provided for achieving balance between channels, employing the aid of a dual magic eye tube, a 6AF6, each half of which is similar to the familiar 6E5. A built-in tone generator is switched to each power amplifier by means of a control marked CALIBRATE. The output from each power amplifier goes to half of a 12AX7 and thence to one of the grids of the 6AF6, causing the eye of each half of the electron ray tube to close partially or fully. The extent to which the eye closes is determined by a potentiometer marked POWER, which is concentric with the switch marked CALIBRATE. There is a set of power and calibrate controls for each channel. The power controls are set to the same position, according to panel markings for 1 watt, 5 watts, 10 watts, 15 watts, and 20 watts. The listener may choose any panel marking he desires, provided both power controls are turned to the same position. This means that each half of the 6AF6 will respond in the same fashion for equal amounts of power supplied by each power amplifier. Then the gain control of the left channel is advanced until the left eye shadow of the 6AF6 barely closes. The gain control of the right channel is similarly advanced until the right eye shadow barely closes. With the eye producing equal indications in each section, there should be equal power output by each amplifier.

The tone controls are separate for each channel instead of being ganged, as is more frequently the custom. For the left channel there is a concentric pair of bass and treble controls; and the same for the right channel. There are opposite schools of thought as to whether ganged or separate controls are most desirable, and this reviewer is on the fence between. Ganged controls make for simplicity of appearance and operation, and since matched speakers are necessary for optimum stereo results, there is a good case for the ganged control. On the other hand, many stereofans will be using different speakers for the right and left channels, at least initially, and in this case separate tone controls may well be desirable in order to allow for the variation in frequency response of each speaker system. Even when matched speakers are employed, it is quite possible that differences in their room location will call for different amounts of tone correction.

Although the Series 320 is rated at 20 watts, it was found that at 1000 cps it clipped at about 14 watts on each channel. This may be due to the use of EL84's in the output tube sockets instead of the 6BQ5A's designated on the chassis. According to the manufacturer, the 6BQ5A's will produce somewhat higher power under the same operating conditions. In any event, the difference between 14 and 20 watts is a matter of only about 1.5 db, which is far from serious. It was found that each channel could turn out a good deal of power at very low frequencies, which is not true of all amplifiers that turn out 14 watts or so at 1000 cps. Thus at 30 cycles, clipping was not observed on an oscilloscope until output reached 12 watts.

The sound of the Madison Fielding Series 320 may be described as sweet and clear. The quality of the sound is confirmed by IM distortion measurements, using 60 and 5000 cps respectively in 4:1 ratio. Measurements for the left channel showed IM of about 0.2 per cent at 1 watt equivalent sine wave power, 0.3 per cent at 2 watts, 0.4 per cent at 3 watts, 0.8 per cent at 5 watts, 1 per cent at 8 watts, 1.4 per cent at 10 watts, and 3 per cent at 15 watts.

Sensitivity of the Series 320 appears to be quite adequate for all inputs. As measured on the left channel at 1000 cps with all gain controls full on, the tuner and tape amplifier (high level) inputs required 270 mv for 10 watts output, the tape head input required 2 mv, and the magnetic phono input required but 3.6 mv. The right channel had about 2 db more sensitivity, but of course the user would correct for this by means of the individual gain controls for each channel.

Equalization appeared to be quite accurate on the magnetic phono input at the low end, being within 1 db of the RIAA curve at 50 cps; at the high end, however, treble cut was some 4 db less than stipulated by RIAA. In the case of a signal taken directly from a tape head, bass equalization was considerably short of the NARTB curve, with only 15 db boost supplied at 50 cps instead of 23 db, using 1000 cps as the reference point. Above 1000 cps the NARTB curve exhibits about 10 db cut out to 15,000 cps, and the Madison Fielding appeared quite accurate in this respect.

It would be unfair to omit from this extensive discussion of the Series 320 some mention of the very handsome cabinet which is available for the amplifier at moderate extra cost. The cabinet is in a natural wood finish. The amplifier complete with cabinet has a low silhouette that makes it attractive and suitable for table-top or booksheld use, assuming adequate ventilation is provided in the latter case.

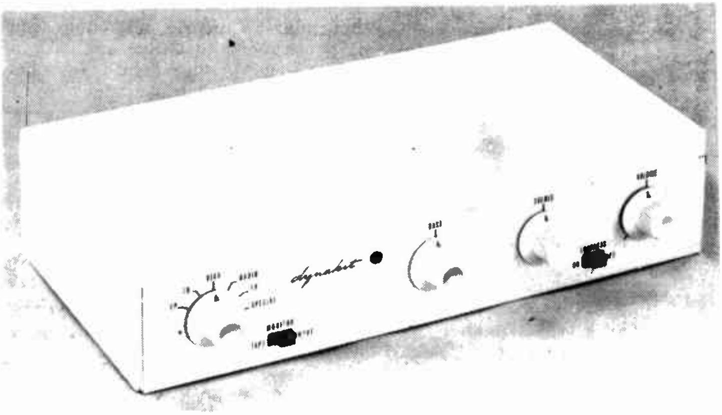
Fig. 1. From a box of parts to a 60-watt amplifier in about two hours, the new Dynakit III is a model any kit builder will enjoy.

DYNACO, INC.

Slightly over a year ago we reviewed the original Dynakit II power amplifier (Sept. 1956) with its unusual—for then—50-watt output. Now there are many 50-watt amplifiers on the market, and some of still higher power, and it is somewhat of a surprise to note how many of them employ the same circuit as that of the Dynakits. The newest model, the Dynakit III, is rated at 60 watts and incorporates a few features that were not part of the earlier model. While the circuit is essentially the same, the output tubes are KT-88's, which accounts largely for the increased output. In addition, a filter choke has been added, improving the signal-to-noise ratio, and a 11.2-ohm resistor has been added from the cathodes of the output tubes to ground—an improvement towards more output linearity that has already been recommended as a suggested change for earlier owners of Dynakit II.

The new unit is the same size as its predecessor—9 by 9 by 6¾ in.—and is of similar appearance, as shown in *Fig.* 1. Rated output is obtained from an input of 1.6 volts, and the model tested (which was assembled in slightly less than two hours) reached 68 watts at 2 per cent IM distortion. Output impedances of 4, 8, and 16 ohms are available, and the unit provides power for Dynakit and Heathkit preamplifiers directly, or for most others if proper connections are made to the power socket.

The Dyna Preamp is an interesting design, conventional in some respects, unusual in others. The tone control action is obtained from a feedback circuit with somewhat less interaction than is usual for the average preamp. One useful feature is the "Special" input which may be wired by the builder to be equalized for microphone or tape head, or it may be employed as an additional RIAA phono position, depending on the user's requirements. In the latter connection, it is possible to leave two pickups plugged in at all times, with front-panel switching—accommodating, for example, a changer and a turntable at the same time. Another very desirable feature is the rectifier-filter circuit which permits the use of a 6.3-volt a.c. source to provide some 11.5 volts of d.c. for heater supply. It should be noted that when this arrangement is used, the ground connection normally present in the power amplifier should be removed, since the ground on the heater circuit is best obtained from the potentiometer in the preamp. The Dynakit III is normally wired with the heater winding free from ground, and if it is to be used with any other preamplifier the center tap should be grounded. Similarly, if the Dyna preamp is used with any other power amplifier, the ground on the heater circuit of the other power amplifier should be lifted. *Figure* 2 shows the external appearance of the Dyna preamp, while *Fig.* 3 shows the internal arrangement. The use of pre-assembled etched-wiring panels reduces construction to a relatively simple operation. In spite, however, of the ease of construction, it is likely that most users would be very well satisfied with the Dyna models.



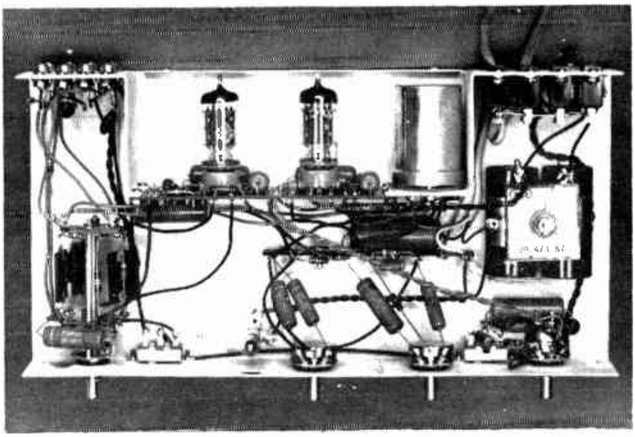


Fig. 2 (left). The Dyna preamplifier kit.

DYNAKIT STEREO CONTROL

The Dynakit Stereo Control (DSC-1) is an inexpensive, easy-to-build kit—one evening's work—intended for adapting two conventional monophonic preamplifiers to stereo use. It is a passive unit (no tubes), so that it introduces no distortion, consumes no power, and has very little that can go wrong. Being a passive unit, it cannot supply signal gain; at the same time, it has very little insertion loss, so that there is little chance of other components having to work significantly harder at higher distortion in order to compensate for such insertion loss.

Primarily, the DSC-1 is meant to be used with preamplifiers that contain a tape-monitor switch. In such a preamplifier, the gain control is connected by this switch either to the selector switch or to the tape input jack (for accepting a signal from a tape playback machine). The tape output jack (for making a tape recording) is linked to the selector switch. The DSC-1 is inserted between the tape output jack and the tape input jack, with the tape-monitor switch in "tape" position, which causes the gain control to be connected to the tape input jack. Hence an incoming signal (tuner, phono, etc.) is routed through the selector switch to the tape output jack, through the DSC-1, into the tape input jack, and to the gain control.

The DSC-1 can also be used between the output of a preamplifier and a power amplifier. In this case, the preamplifier will always operate at a relatively high level, since gain is controlled by the DSC-1 *after* the preamplifier. Consequently, it may be necessary to exercise care that the DSC-1 is operated with the master gain control quite well advanced, allowing the preamplifier to operate at a reduced level in order to avoid significant distortion.

Another alternative, where the monophonic preamplifier does not contain a tape-monitor switch, is for the handy audio fan or a technician to interrupt the signal path between the selector switch and the following stage, and to insert the DSC-1 in this path.

The manufacturer further points out that the DSC-1 can be connected between a signal source (such as a tape playback machine), and a power amplifier, without an intervening preamplifier. In this case one would not have the tone controls and other facilities afforded by a preamplifier. However, a source such as a tape machine or tuner would usually provide at least a gain control.

Despite its outward simplicity, the DSC-1 reflects a good deal of sophisticated thinking about the problems entailed in controlling and coordinating two stereo channels, and it provides in a logical manner most of the stereo functions that are desirable. Let us consider these functions one by one.

1. *Master Gain Control.* The vital requirement of such a control is that it have low tracking error; in other words, throughout its rotation this control should provide about the same attenuation for each channel. Thus if the two channels are balanced when the master gain control is at maximum, then they should stay balanced as gain is reduced. When tracking error is kept within ±3 db—that is, no more than 3 db deviation between channels —it may be considered satisfactory; within ±1 db is considered excellent. Following are the writer's measurements on the DSC-1 he assembled. Using the left channel as a reference at various gain settings, deviations of the right channel with respect to the left channel were as follows:

Reduction in Gain of the Left Channel	Right Channel Gain Relative to Left Channel Gain
0 db	0 db
−5	−2
−10	−4
−15	−4
−20	−1
−25	0
−30	+1
−35	0
−40	−2
−45	−2
−50	−5
−55	−5

It may be seen that the deviation ranged from 1 db to −5 db, which is equivalent to ±3 db. If one were to set the balance control so as to favor the right channel by 2 db, then the tracking error would actually be ±3 db within most of the range of the master gain control.

2. *Channel Reverse.* In addition to serving the usual function of allowing the left channel to be fed to the right speaker and the right channel to the left speaker, this switch facilitates balancing one's equipment for stereo. An accepted technique for such balancing is to feed the same signal alternately to each speaker, meanwhile manipulating the balance control or other controls (such as input level sets on the power amplifiers), until the sound from each speaker appears equally loud. To do so, one may feed a signal into just one channel (from a disc, tape, tuner, oscillator, etc.), and by flicking the channel reverse switch up and down one may hear the sound alternately from each speaker.

3. *Balance Control.* When the balance control knob on the writer's unit was pointed to 12 o'clock, exact balance between channels was obtained, so far as the DSC-1 was concerned. The setting for close balance is not at all critical; very little change in balance occurred between the 11 o'clock and 1 o'clock position of the control.

When the control is turned, say, to the left, this lowers the gain of the left channel without appreciably raising the gain of the right one. The writer measured about 15 db maximum attenuation of the left channel when the control was turned fully counter-clockwise from mid-position, with the gain of the right channel being increased only about 0.5 db. (The same of course applies to the right channel.)

4. *Blend Control.* This is quite a clever device, consisting actually of three controls in one: (a) At maximum counter-clockwise position, the two channels are completely isolated, operating in true stereo fashion. (b) A slight clockwise turn of the knob actuates a switch, introducing a high-value resistance between the two channels, which causes a slight amount of blending; isolation between channels is then about 20 db. As the blend control is turned progressively clockwise, the resistance, and therefore the inter-channel isolation, decreases, eventually approaching zero when the control is almost fully clockwise. (c) When the control is turned completely clockwise, another switch is actuated, causing one signal to be cut out (whether the left or right signal is cut out depends upon the position of the channel reverse switch), while the other signal is fed to both channels. This permits a monophonic source to be fed to both speakers in pseudo-stereo fashion.

The advantages of the blend control—between the points at which either switch is actuated—are at least two: First, it aids in overcoming effect of excessive microphone spacing, excessive speaker spacing, or a combination of the two. Second, at maximum blend position it permits adding the signals produced by a stereo cartridge when playing a mono record, thereby causing the audio signals (lateral information) to add and the vertical rumble (vertical information) to cancel; vertical rumble is ordinarily a good deal more serious than lateral rumble.

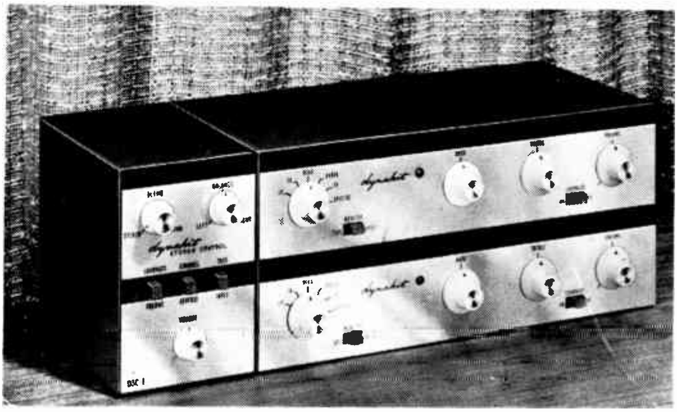


Fig. 1. Dynakit Stereo Control, shown at the left of two Dvnakit preamplifiers, which it matches in appearance.

By turning the blend control from maximum counterclockwise position to maximum clockwise, the experimentally-minded listener can make interesting comparisons among four modes of reproduction: true stereo, blended stereo, both speakers reproducing the same signal containing the total audio information and both speakers reproducing the same signal containing information only from one channel (either the left or the right, depending upon position of the channel reverse switch).

5. *Loudness Switch.* When the DSC-1 is used with two preamplifiers, the gain controls of the latter will ordinarily be at a highly advanced position, thereby allowing the gain control of the DSC-1 to cover a wide range of attenuation. In many or most cases, this will eliminate loudness compensation, which ordinarily takes place when a preamp's gain control is at a reduced setting. Therefore the DSC-1 allows bass boost to be switched in, the amount depending upon the setting of the master gain control. The writer measured about 1.25 db boost at 50 cps when gain was reduced 10 db below maximum; 5 db at 20 db below; 12.5 db at 30 db below; and 15 db at 40 below. No treble boost is provided. However, Fletcher-Munson compensation calls for relatively little, and the treble controls on the preamps are normally more than adequate for this purpose.

6. *Tape Input Switch.* When connected to preamps having a tape monitor switch, the DSC-1 pre-empts their tape output and tape input jacks, as explained earlier. Therefore the DSC-1 provides two substitute sets of tape output and tape input jacks to permit making a tape recording and playing back a recording. Pushing the tape input switch to "tape" permits one to hear the signal from a tape machine. In the "input" position, the DSC-1 feeds through the signal sources to which the preamps are connected.

Being a passive unit, the DSC-1 is a relatively high impedance affair in order to present a sufficiently high load resistance to signal tources. Hence preservation of high frequency response becomes a problem —a problem successfully met in this instance. To minimize losses due to cable capacitance, the DSC-1 is furnished with four low-capacitance cables of one-foot length—two for input and two for output.

With the master gain control full on and the channel balance control at mid-position, the writer found that frequency response was flat to at least 15,000 cps and no more than ½ db down at 30,000 cps. With the master gain control set for 6 db reduction in gain, which is the position entailing the most severe high-frequency losses, response was still flat to 15,000 cps. With gain full on and the balance control set for 6 db reduction in one of the channels, there was about 1 db loss at 15,000 cps in this channel; with the balance control at extreme position (an unlikely state of affairs, involving about 15 db reduction in one channel), the loss was only about 2 db at 15,000 cps. These measurements were taken with the DSC-1 connected to a Dynakit preamp, which itself measured flat beyond 15,000 cps.

It should be added that the DSC-1 is a well-shielded unit, and the writer found that used in conjunction with the Dynakit preamp it introduced no hum problems.

Fig. 5. Heathkit EA-2 amplifier "sounds as good as it measures" and looks as good as it sounds.

THE HEATHKIT EA-2 AMPLIFIER

At $27.95, the Heathkit EA-2 composite amplifier—phono preamp, tone controls, and 12-watt power amplifier, on one chassis, as shown in *Fig.* 5—is a surprise package. Its low price might lead the casual observer to dismiss it as something of a toy, perhaps worthy only of a junior audio system for the children or the den or the workshop, or the like. Actually, it is a grown-up performer, within its 12-watt rating. And it must be remembered that the difference between a 12-watt amplifier and a 30-watter, which few sneeze at, is only 4 db, a relatively slight volume difference to the ear.

The circuit of the EA-2 is essentially orthodox and at the same time up to date. The magnetic phono preamp consists of a single stage, followed by losser type equalization. Subsequently there is a stage of gain for all the inputs, a volume control, another stage of gain, and the tone controls, which are the conventional Sterling type. The power amplifier section follows the trend toward use of a pentode input stage direct-coupled to a triode employed both as a phase inverter and driver for the output tubes. Pentode and triode are in a single envelope, the now widely used 6AN8. The phase inverter is the familiar split-load type, which some experts hold to be as good as any. The output stage is ultra-linear, using the highly-regarded EL84's.

The EA-2 is easy to assemble and took this reviewer the equivalent of four evenings. (The reviewer probably takes a good deal more time than the average, because he checks all resistors for value, checks capacitors for value and leakage, checks continuity of connections by means of an ohmmeter, doubles back after every dozen steps or so to check his work, and so on. On the other hand, the careful approach has paid off in that every one of the dozen or more kits he has built in this manner has worked correctly right from the start.) Although the chassis is only 8-3/16" deep by 12½" wide, the layout is uncramped and at no time requires a surgeon's dexterity.

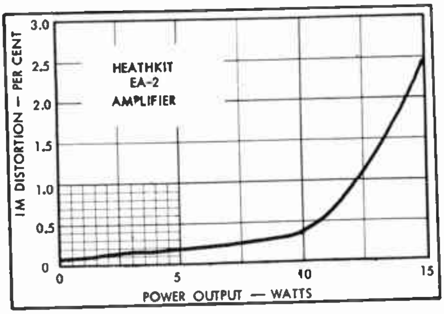


Fig. 6. IM distortion curves for the Heathkit EA-2.

Parts of high quality, as for example the use of molded paper capacitors, an ECC83 as the input tube, EL84's as the output tubes, an EZ81 as the rectifier, and power and output transformers about as husky as can be accommodated within the amplifier's dimensions.

Size of the output transformer is reflected in the ability of the EA-2 to turn out at least 12 solid, clean watts at 20 cps, as viewed on an oscilloscope. At the rated power of 12 watts, the frequency response measured was ±0.5 db between 30 and 10,-000 cps, and 1.5 db down at 20 and 15,000 cps, and 3 db down at 20,000 cps. Below 12 watts, frequency response remained essentially the same within this range.

High-frequency response was also checked with the gain control set at 6 db below maximum; the resistance of the gain pot acts as a low-pass filter in conjunction with the input capacitance of the 6C4 (largely due to Miller effect), and the low-pass action is greatest when the pot is at mid-resistance. At this setting, response was down only 0.5 db more at 10,000 cps, 1.2 db more at 15,000 cps, and 2 db more at 20,000. This slight a deterioration in treble response is hardly apt to be noticed. Moreover, it is quite unlikely that one will be operating the amplifier with gain as far advanced as 6 db below maximum. With gain 10 db below maximum, there was no additional loss at 10,000 cps, about 0.5 db at 15,000, and 1 db at 20,000. At 20 db down, no additional losses were observed.

To obtain maximum flatness of response, with bass and treble control knobs pointing straight up (12 o'clock), it was necessary to rotate the treble control pot about 15° counter-clockwise (center lug pointing to about 11:30). Parts tolerances would account for this. It was not necessary to adjust the mounting of the bass control pot.

Probably the most fascinating thing about the EA-2 is its low distortion. As shown in *Fig.* 6, it does not exceed 1 per cent IM until equivalent sine wave power (the wattmeter reading of two signals mixed in 4:1 ratio is multiplied by 1.47 to obtain the power of a sine wave having the same peak) is above 12 watts. At 10 watts equivalent sine wave power, IM was only 0.33 per cent. From 3 watts down, it measured 0.1 per cent or less. The IM meter employed for these measurements uses frequencies of 60 and 5000 cps and has a residual reading of about .06 per cent. At 10 watts and below, the performance of the EA-2 leaves little if anything to be desired with respect to distortion. And it bears repeating that most audiofans will not be using more than five watts.

An amplifier that measures well with respect to distortion may not sound clean. Often it will be found that such an amplifier displays excessive ringing when square waves upward of 1000 cps are passed through it. The EA-2 exhibited *no ringing whatsoever* on square waves of 1000, 5000, 10,000 and 20,000 cps, and even with treble boost applied.

Still on the subject of distortion, it was found that at 1000 cps the EA-2 clipped at just about 15 watts. So one can think of the EA-2 as a 15-watt amplifier since IM is just a little over 2 per cent at this point.

Sensitivity of this Heathkit is quite sufficient. It was measured at approximately 0.3 volts on high-level inputs and 8.6 mv on the magnetic phono input for 12 watts output. Since one can count on approximately 15 to 20 mv on peaks from even the weakest magnetic cartridges, no problem of inadequate gain is anticipated on magnetic phono input. High-level sources generally turn out from 0.5 to 3 volts on peaks, so there is no problem in this respect either. In fact, the problem may be too much signal input. For example, some piezoelectric cartridges deliver a volt or two on peaks, and some magnetic cartridges put out much as 100 mv or more. In the case of high-level sources, the input signal could easily be cut by reducing the lower leg of the voltage divider at the input. The signal of a high output magnetic cartridge can be reduced by replacing the 47,000-ohm load resistor with an appropriate voltage divider consisting of two resistors having a total value recommended by the cartridge manufacturer.

Only RIAA phono equalization is provided; however, this suits virtually all records presently made and, with slight adjustment of the tone controls, is adequate for records made prior to 1954. Between 30 and 15,000 cps, equalization did not deviate more than 1.5 db from the RIAA curve.

The tone controls of the EA-2 provide a substantial range of boost and cut. Ample bass boost is particularly welcome since no loudness compensation is provided for the Fletcher-Munson effect. At 30 cps, a maximum of 16.5 db boost and 17.5 db cut were measured. Maximum treble boost measured 16 db at 15,000 cps, and cut 21.5 db.

Since the EA-2 aims so high, it is not unfair to talk about its drawbacks, even though its price is so low. One drawback is the provision of only three inputs—tuner, crystal-phono (suitable for ceramic cartridges as well), and magnetic phono. However, a tuner, TV, tape machine, or other high level source can also be fed into the crystal-phono input. The audiofan desiring more inputs probably would not find it difficult to replace the existing selector switch and to mount an additional input jack or two.

A salient omission is an output for feeding a tape recorder. But this could be rectified quite easily, if desired. In fact, in constructing his EA-2, the reviewer paved the way for such an addition in the future. Instead of mounting a seven-pin wafer socket for V2, a 6C4, he mounted a nine-pin ceramic socket in the same hole and employed half of a 12AU7, which is the same as a 6C4. The other half of the 12AU7 can eventually be employed as a cathode follower, requiring only a coupling capacitor and three resistors to be added to the circuit. The cathode follower would be inserted between V2 and V3, and a jack intended for feeding a tape recorder would be connected to the output of the new stage.

In terms of performance, the only criticism that can be directed at the EA-2 is its modest signal-to-noise ratio. On high level inputs the reviewer measured 60 db noise and hum below 12 watts output at 1000 cps. On magnetic phono input he measured a 47 db signal to noise ratio. Ratios at least 10 db higher would be more in line with professional performance.

With gain control full down, slight hum can be heard within a few feet of a speaker of average efficiency. This originates in the power amplifier section (V3, V4, V5) and is likely due to inadequate filtering of the B+ supply for the output tubes. In a quiet listening room and when the program source contains little noise, the hum might be bothersome to a listener sitting close to the speaker. We tried adding a 30-ohm resistor between the rectifier cathode and the B+ lead of the output transformer, bypassing this point to ground with a 40 μf electrolytic capacitor for the additional filtering and increased the signal-to-noise ratio by about 11 db.

But overall, as it stands, the EA-2 provides exceptional performance at its price, and in a number of respects excellent performance at any price. To the handy audiophile, it furthermore offers attractive opportunities for increasing flexibility of performance; and possibly he may find a way to reduce hum, if it does turn out to be a problem in his case.

HEATHKIT W-6M AMPLIFIER

There was a time when the "home-built" amplifier was the only type available to the hi-fi enthusiast, because there were no factory-built models for this market. These amplifiers were also, in most instances, home-designed, and they did not *always* perform as their designers hoped. Now, of course, there are all sizes, types, and colors of amplifiers available as finished products, and all may be presumed to work satisfactorily from the first moment they are plugged in. There are still plenty of people —this observer is one—who enjoy building something, particularly when in doing so we can save quite a bit of money—basically that representing factory labor and its associated overhead and profit. Heath equipment has long been noted for its reliability, and in the new W-6M 70-watt amplifier the results are all that could be desired, and at a price that betokens a considerable saving.

This amplifier, shown in *Fig.* 3, measures 14¼ inches wide, 12½ inches deep, and 9½ inches in height, and has a shipping weight of 59 pounds. Most of this weight is, as would be expected, in the two transformers, so it is obvious that there is no skimping on quality. The circuit, which is shown in *Fig.* 4, offers some innovations which result in a high degree of performance. The first two stages consist of the two halves of a 12AU7, direct coupled. The second half is the usual split-load (cathodyne) phase splitter, and it feeds a 12AX7 voltage amplifier, which in turn feeds a 12BH7 which is a cathode-follower driver for the two 6550 output tubes in an ultra-linear circuit.

The power supply uses a voltage doubler circuit with four silicon rectifiers and more than adequate filtering. An extra winding on the power transformer provides 130 volts to a selenium rectifier for bias supply. Plate currents in the output stage tubes are metered, and provision is made for balancing the two tubes by varying the bias on the driver tubes. Conventional output impedances of 4, 8, and 16 ohms are available for loudspeaker loads, and an additional 70-volt output tap is provided for feeding large speaker distribution systems. When driving loudspeaker loads, the damping is adjustable over a range from 0.5 to 10 by means of a continuously variable control.

Performance

Frequency response is within ±0.5 db from 6 to 70,000 cps, with smooth rolloff beyond these limits to ensure transient stability. Power output is down 3 db at about 13 cps, while harmonic distortion remains below 0.25 per cent over the important ranges, and only reaches 1 per cent at 70 watts at frequencies of 20 and 10,000 cps. Intermodulation distortion reaches 1 per cent at about 73 watts, and at our rating point—2 per cent IM—the output was measured at 81 watts. Full output is reached with an input of 1.1 volts, and hum and noise measures lower than 70 db below 1 watt.

One of the problems encountered with the Williamson-type circuit—comprising the direct-coupled input pair of stages—was its poor performance as regards overload recovery. This was shown by oscillograph traces of signal output when the level was changed quickly from a high value to a very low value—a condition that is common in musical program material. No such instability was observed with the W-6M, however, and only the slightest amount of ringing was noticed on 10,000-cps square waves when driving a loudspeaker load, and none at all on frequencies below 2000 cps.

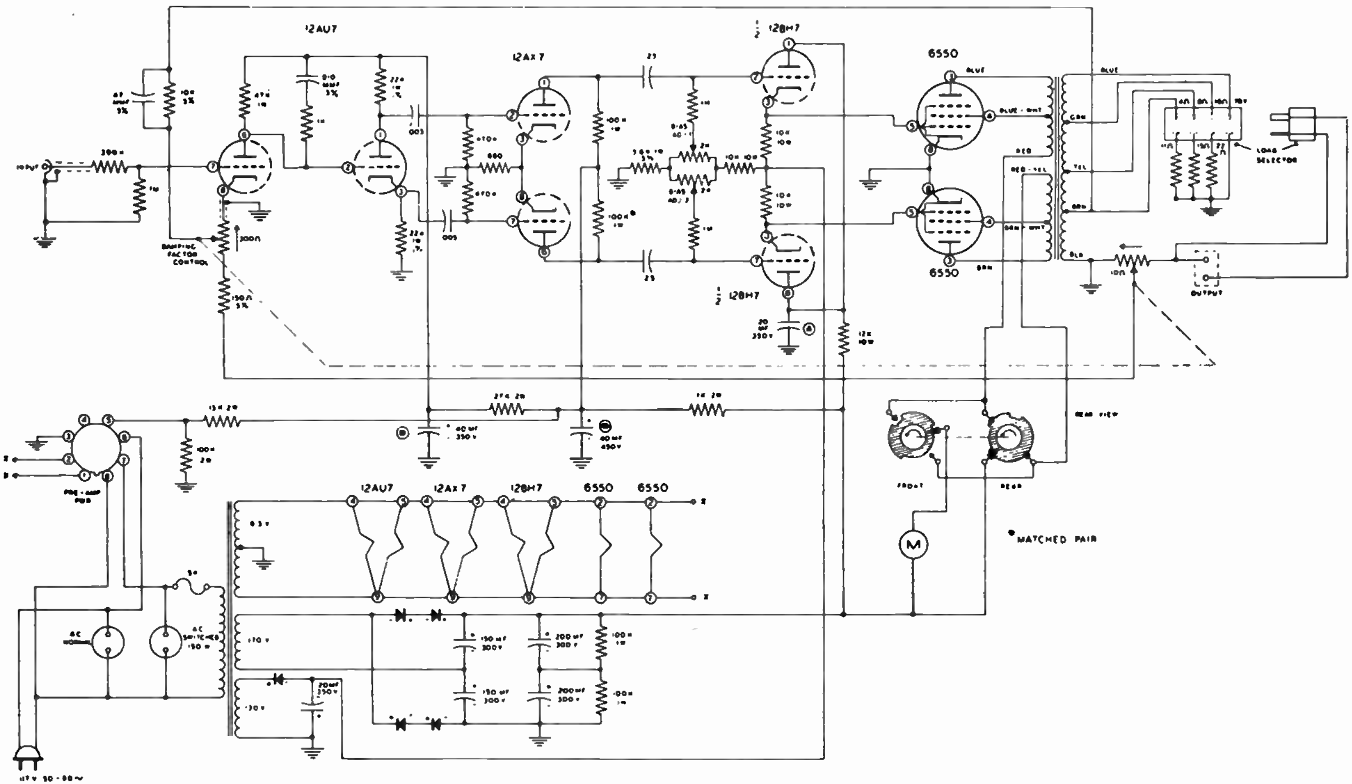


Fig. 4. Over-all schematic of the 70-watt Heathkit amplifier.

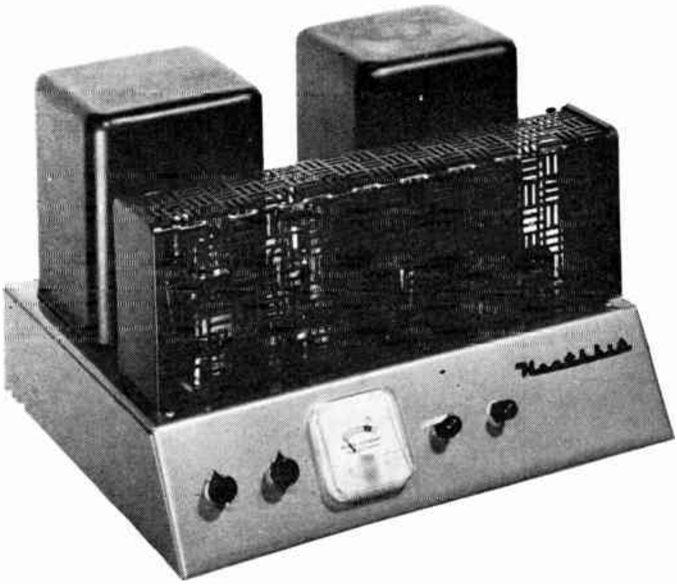


Fig. 3. External appearance of Heathkit W-6M-70-watt amplifier.

Variable Damping

The schematic of *Fig.* 4 shows an unusual arrangement of the output wiring. Note that the variable-damping-factor control is a dual potentiometer, with the 10-ohm section in the return side of the output winding. In order to maintain constant gain as the damping factor control is rotated, three different resistors are used for the three low-impedance output taps. Thus the control—which changes the ratio of voltage feedback to current feedback—can be calibrated directly in damping factor, and gain and distortion remain constant for any setting of the control.

With a high-quality speaker system there is little difference in performance as the damping-factor control is turned, but as the quality of the speaker and enclosure is lowered, the effect becomes more and more noticeable. The higher values of damping factor minimize cabinet resonance and thus reduce any boominess that might result from poor enclosure balance. When used to drive a number of speakers at the same time, the damping factor should best be operated at its maximum position.

Construction

As with other Heathkits that we have had personal experience with, the W-6M "builds" nicely. The instructions are well written, and give the impression that once being completed, they were possibly given to a completely inexperienced constructor to find out if they were sufficiently clear and complete. After completing and testing the amplifier, we "unbuilt" it far enough to add a 25-volt transformer, a full-wave selenium rectifier, and filter capacitors so as to have a 24-volt d.c. supply for a new preamplifier. The space between the power and output transformers is wide enough to accommodate the rectifier and the capacitors, and the extra transformer will just go into the space under the output transformer.

And then—after the manner of silent picture subtitles—came stereo. The problem now is to find space enough (and strength enough) in a cabinet to hold two of these units—118 pounds—completely aside from the need for physical strength enough to lift them. We shall remain quite content with a smaller amplifier for the second speaker, using this model for the principal speaker and the five others that are distributed around our home at strategic locations. By which we mean to imply that we consider this one of the better amplifiers available and will continue to use it.

That is, we suppose until somebody introduces a practical 100-watt amplifier for home use.

KNIGHT-KIT 83YX776 STEREO PREAMPLIFIER

By their very nature, a stereophonic preamplifier must become a rather complicated device, when one considers the number of individual tube circuits involved. About the minimum number of stages required for a monophonic preamplifier employing tone controls is six—two for the phono preamp, two triode sections for the tone-control circuitry, an additional gain stage to make up for the use of both level-set and loudness controls, and a final stage as a cathode follower. Some configurations of circuitry reduce the total number of tubes required to only two—a pentode preamp stage followed by a second pentode which is followed in turn by the tone and volume controls. Others make use of only two double triodes, with four tube circuits. The majority of control units, however, do employ six triode sections, which makes for a simple straightforward design with plenty of isolation between the various control circuits so that there is no interaction. When the circuitry is duplicated for stereo, there are then twelve stages, and anyway you look at it, this means a lot of connections.

The Knight-Kit Stereo Preamplifier—with the unlikely model number 83YX776 (actually a catalog number), employs this type of circuitry, but does it in such a manner that the work involved in putting it together is reduced to a reasonable minimum. We have seen some complicated kits which take as much as twice the construction time as this one. In addition to placing most of the audio components on printed wiring panels, Knight uses rotary switches designed to "plug in" to the ready-made panels, which reduces the work considerably. Instead of making connections to 45 of the somewhat delicate terminals of wafer switches—and with often more than one connection to each terminal—the switches are simply inserted into the ready-punched holes and the projecting pins soldered to the printed wiring. Not only does this simplify the work, but it also eliminates the possibility of making wrong connections to the switch terminals, which can be done easily when as many as 36 or 48 terminals appear on one switch.

Circuit Description

Among its many features, the Knight-Kit Stereo Preamp accommodates five pairs of stereo inputs and four monophonic inputs —the latter being arranged for three different types of pickups and for a microphone. Bass, treble, and volume controls are clutch-type units which permit independent adjustment of each element simply by holding one knob while turning the other, though both may be turned together as though they were conventional dual controls. Sharp cutoff scratch and rumble filters are provided, and to minimize hum, all heaters are operated on direct current.

From left to right, as shown in *Fig.* 2, the controls are: selector equalizer, channel selector, bass, treble, level, and loudness/power. The slide switches are the rumble and scratch filters.

The selector equalizer switch has four stereo positions (tape head, phono, tuner, and auxiliary), and seven mono positions (microphone, and phono equalizations for RIAA, European, 250-cps crossover, ffrr, old AES, and NAB). These monophonic phono positions apply to the inputs labeled GE, PICkering, and CERamic, and designed to accommodate medium and high-level magnetic cartridges and the ceramic types.

The second control from the left is the channel selector, which has six positions—two for stereo and stereo reverse, two pro-

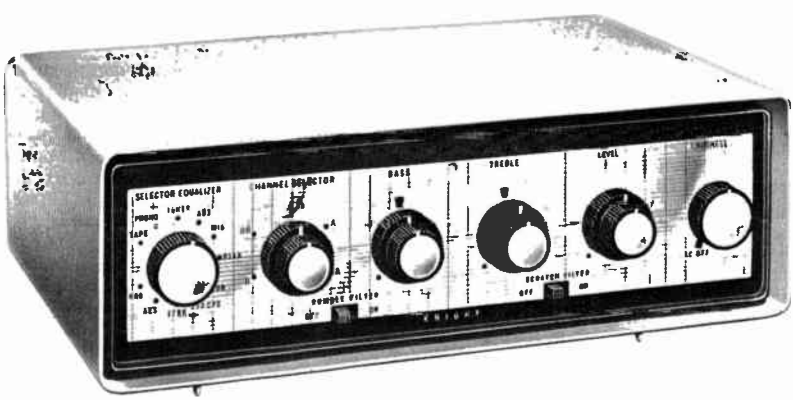


Fig. 2. Knight-Kit Stereo Preamplifier.

vide for individual channel operation with the other channel inoperative, and two provide for outputs on both channels (monophonically) from inputs to either channel A or B. This switch has 360-deg. rotation.

The third and fourth controls are bass and treble, respectively, followed by the level control. These three controls have separate knobs for each channel, and a soft rubber washer between them serves as a clutch and permits either independent or common operation. The sixth control is loudness, and is common to both channels.

The preamplifiers consist of dual triodes with losser-type equalization between stages. These are followed by the tone control pair which feed the level controls, and the arms of the level controls feed the loudness controls. A gain stage and a cathode follower, with feedback around both, come next, with their outputs fed to the channel selector, and its outputs are fed in turn to the scratch filter and to dual output jacks for each channel. The rumble filter works only on phono and tape head inputs, and consists of a two-section RC network in each channel. The power supply consists of a single-secondary transformer with selenium rectifiers, followed by a low-resistance high-capacitance filter for the heater circuit, with all six heaters and the pilot light being fed in series; plate supply is filtered by a conventional RC filter. One a.c. receptacle is provided ahead of the power switch, and another is switched. Line noise is filtered from the primary circuit by two .01-μf capacitors from line to ground. Tape recorder feed jacks connect ahead of tone and volume controls.

Level-set controls are provided for the two tuner inputs, and a calibrated dual control is used to set the phono pickup load on the stereo phono inputs, with a range from 5000 to 105,000 ohms. These three controls are accessible from the bottom of the unit, as is also the line fuse.

Construction

The Knight-Kit Preamplifier is housed in a crackle-finished metal cabinet 13¼ in. long, 8 in. deep, and 4¼ in. high, plus another ¾ in. for the brass legs. The front panel is aluminum, with a "graph-paper" design, and knobs are black with chromium-plated inserts. The back of the cabinet is perforated metal, and in an operating position no jacks or connections are visible from the rear. All connections are made on a small sloping panel in a recessed area in the bottom of the cabinet, so that for table-top use all cables could be brought up through a hole in the supporting surface and none would show anywhere. This would also apply when installing the unit in a bookcase, for example, with the leads carried through the shelf behind the books on the next shelf below.

Aside from the power supply, the input and output jacks, and the volume and tone controls, practically all of the component parts are mounted on the printed wiring panels. There are two of these, one carrying the six amplifier tubes and most of plate and cathode resistors and the coupling capacitors, while the other carries the two switches, and the equalizing-network components. The two panels are assembled separately, and interconnections between the panels and the other elements of the preamplifier are made after the panels are mounted in place. The tubes mount in a vertical position, and are accessible by turning the cabinet over. The power-supply section is housed in a separately shielded section of the chassis.

There are some 355 separate operations to the assembly of this unit, which is less than half of what would be expected from equivalent circuitry if assembled in the conventional manner. There are 106 resistors and 56 capacitors (one is a dual ceramic and two are triple-section electrolytics, which makes 61 capacitors in effect), so many connections are obviously necessary. Our construction time was just under fifteen hours.

Aside from the circuit design, which we consider to be excellently thought out and to provide a high degree of flexibility in switching, the physical layout of the unit seems to be efficient. It is just possible that it might be difficult to service in case of any part failure or in case of a mistake in construction. Fortunately, the instructions are extremely clear, and our test unit performed perfectly from the first time it was turned on.

Performance

Performance measurements made include input levels for a 1-volt output; frequency response curves for the various types of equalization (both stereo and mono inputs were checked), including tone and loudness controls; frequency response of the rumble and scratch filter; crosstalk between channels; and harmonic distortion at a 1-volt output.

While specifications claim a sensitivity of 2.5 mv for tape-head and phono inputs, we measured both at a sensitivity of 1.85 mv for a 1-volt output, and with a hum and noise output of 64 db below 1 volt. Sensitivity of the ceramic phono input was measured at 65 mv for 1 volt output, the same output was obtained from the auxiliary input from a 500-mv signal, and from a 150-mv signal at the tuner input with the level-set control at maximum. On the monophonic inputs, the "GE" jack required a 1.9-mv signal and the "Pickering" jack required a 5.7-mv signal for the standard 1-volt output; the microphone jack required a 25-mv signal for the same output.

As to frequency response, the "flat" inputs (auxiliary and tuner) were flat within 0.5 db from 10 to 100,000 cps. All equalized inputs were within ± 2 db of the standard throughout, and differences between channels were less than 2 db. Level differences between channels did not exceed 3 db anywhere except at the extremely low settings where they reached 6 db in the worst condition. Tone control curves were within 3 db on both channels, and loudness control compensation was within ± 4 db at the measured points—20, 30, and 40 db below maximum output. Hum and noise on the high-level inputs was 82 db below 1 volt. Crosstalk was measured at 42 db at 1000 cps, and 35 db at 10,000 cps.

The rumble filter provides a cutoff of approximately 5.5 db/octave below 100 cps, and the two curves were within 2 db. The scratch filter—which consists of a choke and capacitor circuit—gives a 10 db/octave attenuation beginning at 2500 cps, and both channels were within 4 db of each other at 6000 cps, within 6 db at 10,000 cps. Harmonic distortion was measured only on the "flat" inputs, and was 0.2 per cent at 1 volt output, reaching 1.1 per cent at a 3-volt output signal.

The tube complement includes two 12AY7 tubes for the preamplifiers, with four 12AU7's making up the remainder of the circuit.

At its relatively low price and reasonable construction time, this preamplifier offers practically any feature that could be desired. We might have preferred the addition of a phase reversal switch six months ago, but they do not seem to be so necessary now, since records all seem to be standardized.

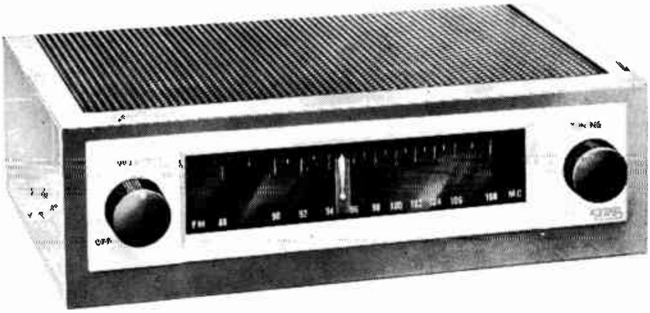


Fig 3. EICO FM tuner—kit or factory built—works right off.

EICO HFT-90 FM TUNER KIT

If we had been told five years ago that it would be possible to construct an FM tuner from a kit and have it work from the first time it was turned on, we would have been properly doubtful, but we now know that it is possible. Not only is it possible, but the performance compares well with factory-built sets in many respects, and after giving the completed kit a check-up with an FM sweep generator, only a very minor improvement was obtained.

Since we insist on actually building any kit equipment we review—so as to be able to assess the clarity and completeness of the instructions—we assembled the EICO HFT-90 entirely from the manual and without attempting to analyze each step as we went along. This is what the inexperienced builder would do, and if instructions are not adequate, he should be so advised. The EICO manual is simple and complete, and the finished set is neat in appearance and excellent in performance.

The specifications claim 1.5 μv for 20 db quieting, full limiting from 25 μv, frequency response uniform from 20 to 20,000 cps within ± 1 db, an i.f. bandwidth of 260 kc, and a peak separation of 600 kc at the detector, a broadband ratio-detector type. In comparison to other sets of known sensitivity, the EICO appears to meet its specifications; the alignment generator proves the bandwidths. So it must be conceded that factory alignment of kit parts *does* work.

The circuit employs an ECC85 in the front end—one half acting as a grounded-grid r.f. amplifier and the other half as a reflex converter. There are three i.f. stages, the ratio detector, and a cathode-follower output tube, together with a unique tuning indicator which is also the dial "pointer."

This subminiature tube has a blue glow pattern which is in the shape of an exclamation point (we wondered if this was intended to indicate surprise that the home-built kit worked) in which the top portion grows smaller as a station is tuned in. This tube is carried on the dial indicator and so serves to indicate where the set is tuned. Adequate r.f. bypassing is provided in the heater circuits, and a 6X4 serves as the power rectifier. The completed tuner, in its ornamental perforated cover, is 3⅝ in. high, 12 in. wide, and 8¼ in. deep. Two outputs are provided—the normal audio output from the cathode follower and controlled by a volume control, and a multiplex output ahead of the de-emphasis network.

Following instructions explicitly, it took about six hours to complete the tuner, including mounting the enclosure. After completion, and with no checks whatever, the set was connected to an amplifier and turned on. It worked from the start, giving excellent reception. To determine how well factory alignment worked, the set was checked with a sweep generator and oscilloscope and only the tiniest improvement could be noted. Over-all impression—extremely good in both audio quality and appearance. Construction—simple and straightforward. Instruction manual—very thorough and accurate.

On the whole, an excellent kit for anyone needing a compact tuner of good performance—and at a very attractive price.

ACROSOUND ULTRA-LINEAR II

For those who like the work of assembling kits but prefer as little work as possible, the new Acrosound Ultra-Linear II amplifier kit is one of the answers. With an output of 60 watts rated continuous power at an IM distortion of less than 1 per cent, this is an amplifier of highest quality. Combined with a variable damping factor control ranging from 0.5 to 10 and a sensitivity of 1.57 volts (measured) for the 60-watt output, the unit will serve with any good preamplifier, and two in a stereo system readily show how much better stereo is when one does not make compromises with power output. While it is agreed that two 10- or 12-watt amplifiers in a stereo system are better by far than one of the amplifiers alone in a monophonic system, there is no question about how much better a system sounds with more powerful amplifiers.

The UL-II employs a 12AX7 as an input stage operating as a "long-tailed pair," with the cathodes returning to about −40 volts, the plates working at 101 volts and directly connected to the grids of the second stage, a 12AU7. This stage has an 18k-ohm common cathode resistor, which makes it also a long-tailed pair. Between the two, the phase splitting is just about perfect, since one tends to equalize the other—with both being aided by feedback from the secondary of the output transformer in a hybrid arrangement as described in the September, 1958, issue. The output stage uses a pair of EL34's, with a GZ34 providing high voltage and a selenium rectifier providing the negative voltage for bias and for the return on the first-stage cathodes.

Construction

The amplifier is of rather unusual physical design, since the chassis is composed of the four side aprons which are held together at the corners and by the two transformers. The open space in the center is occupied by a printed circuit panel on which are mounted all the tubes and all other circuit components except for two resistors and the variable damping control. *Figure* 1 shows the finished unit from the control panel side, *Fig.* 2 shows the amplifier from the underside with the bottom plate removed, and *Fig.* 3 shows the printed-circuit unit by itself. There are 13 mechanical operations in the preassembly of the front panel apron, 12 wiring operations on the front panel, 5 final assembly operations, and 24 final wiring operations—then the amplifier is ready for testing. The entire printed circuit panel comes already assembled and tested, and the kit builder has only to connect leads to the panel and insert the tubes. The entire construction operation should not require more than two hours by the most inexperienced. A little extra time might be needed to connect the power supply socket to accommodate your particular preamp, and instructions are supplied for the Eico HF-61 and the Heathkit WA-P2, which indicate the general type of connection required for any preamp. The unit will furnish power up to 1.5 amps. at 6.3 volts and 20 ma at 485 volts, which indicates that a series resistor would likely be necessary to drop the plate voltage to a more suitable value for the average preamplifier.

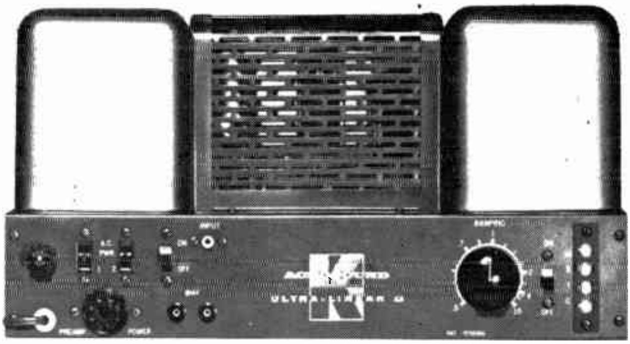


Fig. 1. Acrosound Ultra-Linear II amplifier constructed from a kit.

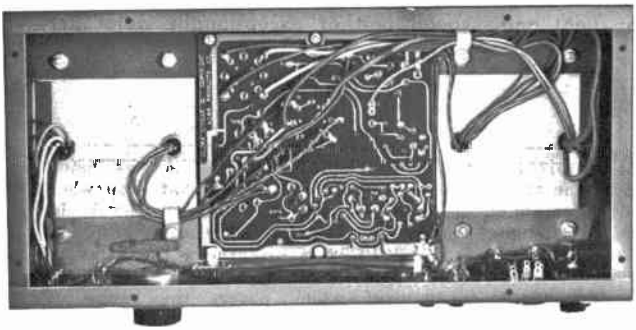


Fig. 2. Bottom view of the Acrosound amplifier showing simplicity of wiring, since the printed-circuit panel is furnished completely assembled.

Performance

The output transformer, Acrosound TO-600, is designed with a separate feedback winding and thus isolates the load on the amplifier and the feedback circuit so that feedback stability is of a high degree. This arrangement also provides for the variable damping feature without necessitating a dual control to maintain constant gain regardless of damping-control setting. The circuit is arranged with three separate operating adjustments—bias on the output stage, balance between the two output tubes, and an a.c. balance control in the second stage which aids in reducing IM distortion to a minimum by dynamically balancing the driver and output tubes. Without suitable IM test instruments, this control is normally set at midpoint, and distortion is in the vicinity of 1 per cent at 60 watts. However, if one has access to IM testing equipment, this control may be reset for minimum distortion, which is claimed to be about 0.4 per cent at 60 watts.

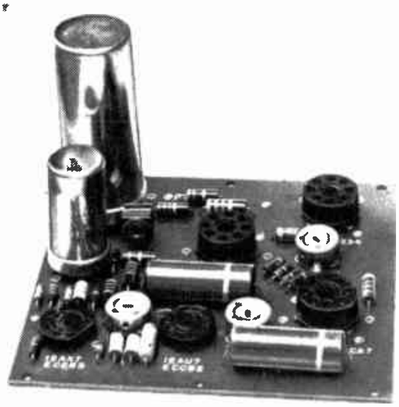


Fig. 3. Printed circuit for the Ultra-Linear II is completely wired and tested as received by the kit builder.

Using 60 and 7000 cps in a 4:1 ratio, an IM distortion of 0.9 per cent was measured with the control set at the midpoint and at an output of 60 watts; adjustment of the control for minimum distortion resulted in a figure of 0.47 per cent on the unit tested.

In addition to being extremely easy to construct, the Acrosound Ultra-Linear II is handsome in appearance and lives up to its specifications. The variable damping control is of no great help with speakers of the best quality, but with those in poor enclosures the boominess can be controlled quite readily. So for either the best or the poorest loudspeakers, this must be rated as an excellent amplifier.

Fig. 7. Garrard Model T "Mark II" 4-speed manual record player—neat and compact.

Fig. 8. The Garrard "Professional" model 301 transcription turntable—a three-speed unit of exceptionally quiet performance.

THE GARRARD LINE

It has long been the custom of this department to review equipment of the type that the average reader considers "standard" in that it rarely seems to change appreciably from year to year. This description applies particularly well to the Garrard line of record players and changers—as well as the 301 transcription turntable—because such minor improvements as may be introduced into the various units are not sufficient to warrant a new model designation. While the Model T "Mark II" was reviewed in July of this year, and the new TPS-10 arm was described in the October issue, the remainder of the line has not been discussed since the introduction of the RC88, RC98, and RC121 models over a year ago.

Figure 7 shows the Model T, which is a manual record player with four speeds. The pickup shell is the same as those used on the three changers, which simplifies the problem of the user who may have a changer in his home but who occasionally uses the smaller manual unit as a "portable", for example. The motor starts when the arm is moved fully to the right, and shuts off automatically at the end of the record. An arm rest is provided, and the entire unit is quite compact.

The RC88—which might be described as the common garden variety since these are the Garrard models most often encountered—is shown in *Fig.* 9. This model uses the pusher platform type of record change for 10"- and 12" records, and a center-drop spindle for the 7" 45's. One interesting and useful feature is the spring mounting used for all of the changers as well as for the Model T. To make installation, the mountings are attached to the base plate and simply pressed into the holes on the motor board, levelling screws being accessible from the top so that even when the motor board itself is not level the changer can be made so readily.

Model RC98 is essentially the same as the 88 with the addition of a vernier speed adjustment which covers the range of about ± 4 per cent, permitting those who wish to play records with a piano or other instrument to match the pitch properly. The underside of the 98 is shown in *Fig.* 11, and it is identical with the 88 except for the vernier speed control housing shown at the left.

Model RC121, shown in *Fig.* 10, uses the simpler center-drop spindle for both types of records. This results in a smaller base plate that can usually be accommodated in the space available in existing phonographs when a change to a better quality of turntable is desired or when the user feels that he wants the 16⅔ rpm speed. Like all the other models, no belts are used between motor shaft and idlers, but a more conventional turret drive is employed to effect the speed changes.

Figure 8 shows the top view of the Model 301 transcription turntable, which has already established an enviable reputation for itself. Using the same basic drive principles, the transcription model is naturally constructed with every possible thought being given to the reduction of rumble. The underside view of this model shows the "battleship" type of construction, with the motor being supported in a cradle by means of rubber-damped springs, and the vernier speed control isolated from the motor by springs to minimize any transmission of motor vibration—and there is very little from this dynamically

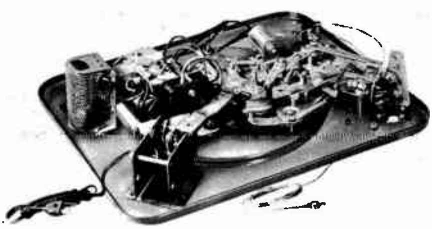
Fig. 10. Model RC121 is more compact because of the center-drop spindle, can fit into many existing phonographs.

Fig. 9. The RC88 record changer—four speeds, automatic or manual operation.

balanced motor—to the chassis. An interlock prevents changing speeds when the motor is running, and the ON-OFF switch control is separate from the SPEED SELECTOR to eliminate the possibility of turning the unit on to a wrong speed—a disaster in professional applications.

As the leader in the Garrard line, the 301 well deserves its excellent reputation.

Fig. 11 (below). Underside view of the deluxe RC98. The vernier speed control—which changes the electrical circuits of the motor itself—is in the perforated housing at the left. The RC88 is essentially identical except for this control. Fig. 12 (right). Underside of the 301 transcription turntable. Note the extremely rugged construction with thorough isolation of motor from chassis.



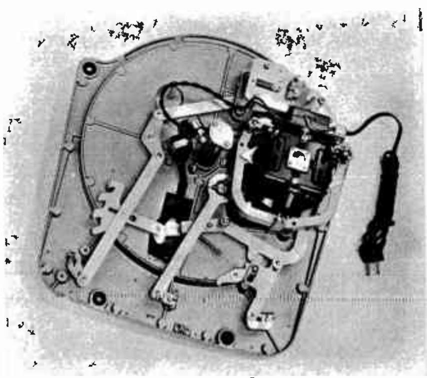


Fig. 5. General Electric "Golden Classic" Stereo Pickup Cartridge.

GENERAL ELECTRIC STEREO CARTRIDGES

With three different models of stereo cartridges on the market, it appears that General Electric covers the entire field. The "Golden Classic" type GC-5 is designed for top performance in transcription type phono arms, and is fitted with a 0.5-mil diamond stylus. This model is intended to work at a tracking force of two to four grams, and the specifications claim a lateral compliance of 4×10^{-6} cm/dyne and a vertical compliance of 2.5×10^{-6} cm/dyne.

Model GC-7, slightly less in price because of the stylus assembly, employs a 0.7-mil diamond, and is intended for tracking forces of 3.5 to 7 grams. The compliances are slightly less than in the GC-5, being specified at 3×10^{-6} cm/dyne for lateral and 2×10^{-6} cm/dyne for vertical. Model CL-7 is the same in all particulars except for the stylus, which is a 0.7-mil synthetic sapphire. Actually, all models are identical except for the stylus assembly, and any stylus may be used on any model, since they are interchangeable.

All are intended to work into a load of 100k ohms for flat output, and the nominal output signal is 6 mv for a stylus velocity of 5.5 cm/sec. Later models have an 8-mv output, we are told, but have not yet had an opportunity of checking.

The physical construction of the GE stereo pickups is similar to that of the VR-II. The two coils are mounted vertically at the forward end of the unit, with their cores extending to form pole pieces which are bent at the tips to form a V, open at the bottom. The tip of the stylus arm is in juxtaposition to the pole tips, and its movements are translated into varying reluctances in the two cores, with varying voltages induced in their surrounding coils. Four terminals are brought out, eliminating one of the problems of stereo pickups—that of serving as a common ground for two separate amplifier systems. The "ground" terminals of the two coils are strapped together at the back of the unit by an extension of the shield, but the shield is scored so that it may be separated readily to provide a four-terminal operation when necessary, or it may be left as supplied if the installation will operate satisfactorily with three terminals.

The principal problem with the GE cartridges is, however, induced hum from external fields. When used with a system with good low-frequency response, the hum is likely to be objectionable with any record changer we have tried so far; we noted no trouble with two different transcription turntables. With the two coils paralleled by shorting the two "hot" terminals, as they would be for monophonic reproduction, the coils are in a humbucking configuration, and there is no noticebale hum. As we pointed out in our construction article last February with the VR-II, when the coils are used in stereo, there is no hum-bucking action. One cure we worked out is to connect a low-inductance choke (0.5 H, for example) between the two "hot" terminals, and this reduces hum by about 12 db without appreciably affecting stereo separation. A choke recommended for this purpose is made by Aladdin Industries, number 18-476, and may be obtained from most jobbers. Any choke used in this application must of itself be kept out of a.c. fields, but the Aladdin units are well shielded and not troublesome in this respect. Such a cure is recommended in severe cases of induced hum.

FAIRCHILD

In general it may be said that the simplest form of any particular device might well be the best, since there are likely to be fewer parts and consequently less chance of trouble. Thus the simplest form of a phonograph turntable could well be the platter with a suitable bearing, a driving motor, and a belt drive between them. This permits the motor to be mounted as flexibly as possible, and removes any actual mechanical contact between motor and turntable, since the belt would undoubtedly be made of some material which would not transmit vibrations.

Such a device is the Fairchild 412-1 turntable—a single-speed model for those whose principal interest in records centers

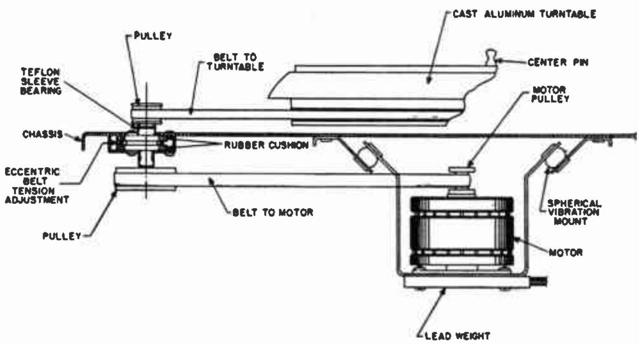


Fig. 4 The Fairchild 412-1 turntable mounted in a hardwood cabinet which accommodates also the 280 transcription arm. Entire unit is large, but a belt drive from motor to idler and another belt drive from idler to turntable holds rumble to a minimum. Fig. 5 (above). Cross-sectional view of the 412-1 turntable showing relative position of the various elements.

around the LP. However, for those who need the other speeds, the Electronic Drive unit may be employed. Basically, therefore, the turntable unit consists of the platter, a motor, and a belt—although, actually, to obtain suitable speed reduction there is an intermediate idler, with two belt drives. One belt drives the idler from the motor, and the other drives the turntable from the idler. *Figure* 4 shows the complete unit, together with the transcription arm, and *Fig.* 5 shows the mechanical arrangement in cross section.

The driving motor—a hysteresis-synchronous model—is mounted in a flexible cradle with highly compliant suspension which eliminates transmission of vibration (and audible noise) to the chassis. The idler is mounted to the chassis with rubber cushions, and is equipped with Teflon sleeve bearings, the whole idler assembly being mounted in an eccentric which permits adjustment of belt tension. The recognized speed constancy of the hysteresis-synchronous type of motor ensures extreme speed accuracy at the turntable proper—

the production models being held to an absolute speed accuracy of closer than 0.3 per cent.

The turntable platter is a heavy aluminum casting, dynamically balanced, and its hardened steel shaft is carried in lapped well provided with lubrication grooves, the thrust being taken on a hardened steel ball riding on a Nylon seat. Rumble was so low as to be extremely difficult to measure, since even the unmodulated grooves on most test records show some "rumble" when the reproducing equipment finally becomes good enough. Suffice that rumble is entirely *inaudible* with any records we played at well above normal listening level.

For those who need other speeds, the Electronic Drive unit may be added at any time. This unit simply provides other frequencies of supply voltage by means of an oscillator—60 cps being normal for the 33⅓-rpm speed, and thus equivalent to normal line supply frequency. For 16⅔ rpm, the oscillator supplies 30-cps drive voltage; for 45 rpm the supply frequency is 81 cps; and for 78 rpm the frequency is 141 cps. Since the absolute speed of the hysteresis-synchronous motor is dependent on the frequency of the supply voltage any speed desired can be obtained by changing the frequency of the oscillator. This makes an especially desirable unit for use in areas where 60 cps is not the normal line frequency or where the frequency is not carefully controlled. The Electronic Drive unit can also be used from practically any voltage source—a.c. from 85 to 135 volts at from 40 to 400 cps; d.c., with any type of inverter, which does not need to be frequency-controlled; or batteries, with an ordinary vibrator-inverter. The ED unit is normally used with four fixed output frequencies, all of which have vernier adjustments, but the output frequencies could be tailored by any user having special requirements with relatively little trouble.

The turntable unit requires a clearance of 6⅞ in. below the mounting board, and the hardwood base shown in *Fig.* 4 is 21¼ by 14½ by 7½ in., actually not much larger than the turntable unit itself. The top board is mounted with acoustic isolation to minimize the effects of external vibration.

The 280A transcription arm is an improved model of the earlier Fairchild arm, and has a minimum of bearing friction—both vertically and horizontally—low vertical mass, and a detent which holds the arm fixed in the rest position without any additional arms or holding devices. The cartridge is carried on a removable slide to which electrical contact is made by a pair of springs which short out the leads when the cartridge slide is removed. Arm resonance has been held to a minimum, with no effect whatever being noticed above 12 cps. The arm is an attractive unit, and well complements the high quality of the turntable.

New Miracord XS-200 record changer—

The advent of stereo has naturally brought with it a whole new complement of amplifiers, pickups, speakers, and other devices made necessary by the additional channel. Such items as might be considered equally usable with both monaural *and* stereo systems have not made any great changes—with the possible exception of adding the word "compatible" to the original description to try to get over the idea that they would work equally well with both.

And while the record changer is one item which should work just as well with either system, it just happens that one of them is the first "dual service" component to show an improved design which is intended to give better performance on stereo, but which at the same time will give improved performance on monaural systems.

The new XS-200 Miracord combines the thoroughly proven qualities of the original XA-100 with some new features which practically put it into the "turntable" category, yet it retains all of the automatic features. Now, instead of a changer which may be used manually at will—thus simulating the professional-type turntable—the new model is described best as "a turntable with record changing facilities." Actually, this is what many people have wanted for years.

Foremost in the modifications is the use of a four-and-a-half pound platter, with the diameter increased to 10 inches. The outer rim of the table is polished and with the almost perfectly true motion it looks—when running—just like the more expensive professional-type single-play turntables. Since the platter is made of cast iron, hum pick-up is reduced considerably with those pickups which are susceptible to external fields. The measured reduction with one of the more popular pickups was approximately 7 db.

Fig. 1 (below). The new Miracord XS-200, with cast iron turntable for minimum rumble, less hum.

The external appearance of the new model, *Fig.* 1, has been made more attractive by the change in color to a two-tone blend of walnut shades, which will go well with most wood finishes. All who have seen the new color scheme have agreed on this point.

While all previous Miracords have been all ready for stereo as regards wiring from the pickup to the terminal strip, (two wires and shield have been standard since the introduction of the XA-100 in this country) a second output lead has been added to the latest version to accommodate stereo cartridges. Thus one has only to plug the two leads into two amplifiers and he is ready to play stereo records—as soon as he has them. If the user wishes to operate the changer with a monaural pickup and amplifier system, he inserts one of the pin plugs into a shorting cap which is furnished, thus connecting one of the "hot" leads to ground. The two small pins on the inside of the head are the "hot" leads, and the large pin is the grounded common, connected to the shield.

The changer operates at four speeds, of course, and the motor has been "beefed up" so that even the heavier turntable attains full speed (at 33⅓ rpm) in less than half a revolution, and rumble is some 4 to 6 db less than heretofore because of the greater turntable mass. Wow and flutter measure between 0.15 and 0.2 per cent, which makes a piano record sound practically perfect.

Center-drop changer spindles are used for all record types—the "Magic Wand" for those with the small center hole and the conventional 45 spindle for those with the 45-type holes. A short spindle is used for manual operation, simplifying the placing and removal of records. In all, this new model is attractive, both in appearance and in performance.

Fig. 2. Tandberg's new model 5-2 is similar in appearance to the earlier models, but offers four-track stereo performance.

TANDBERG MODEL 5-2 STEREO FOUR-TRACK TAPE RECORDER

Some months ago we were called upon to give a talk on stereo, and because of a certain amount of physical efficiency on our part (that's laziness to you) we cast about for a machine that was light enough to be called portable. In many instances the presence of handles on the carrying case is sufficient to warrant that name, but the Tandberg in its case weighs only 27 pounds. We had previously been acquainted with the Model 3-Stereo, but the newest Model 5-2 was still strange to us. *Figure* 2 shows its appearance with its case.

To begin with, this unit is fitted for four tracks, and for three speeds. Thus it will accommodate the promised four-track tapes —removed from the magazine/cartridge and respooled on conventional reels. In addition it provides for extra long playing time at the 1⅞-ips speed. It is entirely self-contained for monophonic recording and for mono or stereo playback, but requires the addition of a second amplifier for stereo recording. This unit is 2¼ in. wide, 8¾ in. long, and 5¼ in. high, and in use is placed alongside the recorder at the left end. It is fitted with a male power receptacle and an output cable which plugs into a receptacle on the recorder head cover. A short power stub is coiled up in the recorder's cable compartment, and furnishes plate and heater power to the auxiliary amplifier when it is in use. The unit accommodates microphone and high-level inputs, and matches in performance the amplifier built into the recorder. A gain control is provided, as are an equalization switch and a level indicator tube.

The recorder itself is a marvel of compact design, and it appears as though each part were made for it, rather than being chosen from usual parts manufacturer's stocks.

There are two complete amplifiers, from tape head to output transformer, each having a 4-watt output. A monitor speaker—5½ by 8 in.—is built in, as is the necessary power supply. In the record mode, one of the output tubes becomes the high-frequency oscillator. The input stages of both playback amplifiers have d.c. on the heaters for a hum level measured at 58 db below the maximum recording level (defined as the 4 per cent distortion point).

Frequency response at 7½ ips was measured as within ±1 db of the Ampex standard tape No. 5563, and with signals recorded and played back the response was ±3 db from 30 to 15,000 cps. At 3¾ ips the output was 3 db down at 10,500 cps, and at 1⅞ ips it was down 3 db at 5300 cps. Flutter and wow at 7½ ips was below 0.15 per cent as nearly as we could measure it—which is not an easy trick, incidentally, when it is so low.

In addition to the low-impedance outputs for direct connection to speakers, a switch on the chassis connects the outputs to the cathodes of the stage preceding the output tubes, thus providing a higher-impedance output, without the potentially present distortion of the output stage pentodes. The level at this connection is around 0.7 volts. For our purposes, we fed the low-impedance signal from the output stage direct to two external power amplifiers which gave a possible 50 watts each for auditorium use. The cathode follower output was not sufficient to drive the external amplifiers, although it would be adequate for insertion at the AUX input of any stereo preamp.

A speed control knob at the top between the reels selects the desired speed and changes equalization at the same time. Just above this switch is the speaker control, which connects the internal speaker to either of the two channels or disconnects it entirely. To the left of the lower head cover is the main circuit control switch, with positions for record, playback, and public address. In the latter position the microphone is fed through to the loudspeaker outputs directly. The volume control knobs are dual, with a friction clutch causing both to turn at once in normal use, but balancing may be done by displacing one with respect to the other. Under this dual knob is a lever which selects the mode of operation—stereo, or mono tracks 1–4, or mono tracks 2–3. The indicator eye is just above the volume control knobs, and to its left is a bass-lift switch, which increases lows by 12 db at 70 cps.

The mechanical operation is controlled by a single gearshift-type knob at the right. For recording or playback the knob is pulled forward; for rewind it is pushed to the left; for fast forward, to the right.

All inputs and outputs except the microphone, which plugs into a jack on the top panel, are available at a terminal panel at the rear. A small compartment is provided for the power cord.

From this description one might surmise that the machine is almost ideal—and so it is, for every use to which we have put it, at least. Starting and stopping is smooth with no spillage of tape, provided the operating knob is pulled forward slightly to start the motor before engaging the idler wheel against the highly polished capstan, and one learns to do this automatically in a very few minutes.

We have used the machine for stereo playback, for dubbing from another machine, and for long-playing background-type music, and so far we have no faults to find with the machine. For any semi-professional or home use we would consider it ideal.

TELECTRO SERIES 900 TAPE DECKS

With the increasing use of preamps which are equipped with tape-head inputs, more and more music lovers are taking a good look at tape decks—as contrasted to complete tape recorders—for their home installations. One of the newest to appear on the market is the Telectro Series 900, which offers a variety of facilities in a simple tape-transport mechanism without electronic equipment. For those who wish a complete recording system, including record and play amplifiers and the necessary bias oscillators, the Telectro line includes the Model TRP-11 record/play amplifier, which provides for recording from low-level microphone input or from a high level source such as a tuner, and in the play mode has an output of approximately 5 volts which is adequate to drive a power amplifier directly. Controls include a record/play selector, equalization switch, noise balance, and gain control, and recording level is shown by a VU meter. Also available is the Model TP-12 play preamplifier, which is similar to the playback portion of the TRP-11.

The tape transport itself is obtainable in five forms, depending on the head complement. Model 900-1 is equipped for monophonic recording and playback and for 2- or 4-track stereo playback; 900-2 has three heads—monophonic erase and play/record heads, and a 4-track stereo head which may be used as a monitor during monophonic recording, or for playing back both 2- and 4-track stereo tapes; 900-3 has two heads, stereo erase and 4-track record/play; 900-4 is equipped only for playback, and has a single 4-track head which will play mono and stereo tapes; 900-5 has three stereo heads, making it possible to monitor a tape during recording. All models require amplifiers; when monitor facil-

Fig. 5. Telectro 900 Series tape transport deck.

ities are provided, it is necessary to have playback amplifiers in addition to any recording amplifiers that may be necessary in order to avoid a lot of plug changing.

Physically, the Telectro decks consist of a 12 x 13 in. motor board which is covered by a dress plate which measures 13 x 14¾ in. This plate is satin finish stainless steel with slightly beveled edges, giving a professional appearance. The head and mechanism covers are molded plastic in a gray color. Five piano-type keys control the operation—STOP, REWIND, WIND, PLAY, and PAUSE. Above the mechanism cover are located the a.c. power switch, the speed control, and the digital tape counter. All models are arranged for 7½, 3¾, and 1⅞ ips. The mechanism employs a single motor, with belt drive to the reel hubs and to the capstan, the latter having a large flywheel for speed stability. The braking system is a single cord which rides in pulleys, and it results in an extremely smooth and effective braking action. In switching from fast wind or rewind to play, we could not perform the operation quickly enough to break the tape. Furthermore, brake action is sufficiently gentle that double-play tape is not stretched or broken. The deck requires a clearance of 1¼ in. above the motorboard, and 6½ in. below. The unit is said to be mountable either vertically or horizontally, but in the model tested there was no provision for retaining the tape reels on the hubs and we would not be satisfied with risking some of our unreplaceable tapes unless we were reasonably sure that the reels wouldn't fall off and wander around the room. However, a tilt of some 10 deg. should be sufficient to keep the reels in place. A solenoid-operated automatic stop releases the transport mechanism in the absence of tape in the slot, and does not depend on metallic strips on the tape.

Performance

In tape handling, the mechanism proved efficient and relatively gentle. Fast forward and rewind time was measured at 1 minute 45 seconds for a 1200-foot reel of tape. The model tested was 900-3, and flutter and wow (at 7½ ips) was measured at 0.2 per cent, and in a 7-minute time test, the absolute speed was 6 seconds fast, which is within 1.5 per cent. Feeding the output of the heads to the TAPE HEAD input of a Pilot SP-215 preamp gave adequate output with the volume control at about one-quarter rotation, and at a measured signal-to-noise ratio of 53 db. Frequency response from Ampex Standard Tape #5563 played within 2 db of the 1000-cps level from 50 to 10,000 cps, and on tapes of our own recordings which have been checked against a professional machine, response was 3 db down at 15,000 cps. This testifies to the quality of the playback heads.

For installation in a home system where an attractive three-speed unit is wanted, the Telectro 900 Series appears to be well built, and to offer ease of operation together with good reproduction quality.

Ampex A-122 Magnetic Tape Recorder-Reproducer and A-692P Amplifier-Loudspeaker

AMPEX A-122 RECORDER-REPRODUCER

TAPE RECORDERS have been with us for a long time—comparatively speaking—and most of the "rough edges" have been rubbed off so that one no longer needs a degree in electronic engineering to make and play acceptable tape recordings. In the process of converting a semi-professional machine into a reliable home instrument that anyone in the family can operate, Ampex has rung the bell.

The A-122 is so easy to operate and the results obtained are so consistently reliable that the tape recorder can now take its place as another music source on a par with the record changer as an important component of any home music system.

The A-122, *Fig.* 1, is only one of a series of machines which employ basically the same chassis but differ in head arrangement and housings. Models A-111, A-112, and A-113 half-track monaural machines housed, respectively, in a tabletop furniture cabinet, a portable case, or a protective grill for mounting in another cabinet. Models A-121, A-122, and A-124 are in the same housings, respectively, but are equipped with in-line stereo playback heads and an additional amplifier so as to play stereo tapes in addition to recording and playing monaural tapes. Because of the great curiosity about the circuitry of stereo machines, the schematic of the A-122 is shown in *Fig.* 2. In order to conserve space, the power supply section has been omitted. It is quite conventional, however, except that it also furnishes d.c. for the heaters of V_1 and V_2. Plate supply to the bias-erase oscillator and to the output stage of the record amplifier is cut off except when recording.

The unit has a four-position switch to control its operation; in position 1 the a.c. switch is open and the INPUT THRU jack (which is normally connected in parallel with the INPUT RECORD jack and to a radio tuner or to two outputs of the tuner when two are used) feeds both channel outputs. Position 2 is MONITOR, and maintains the feed-through connection, but with the power on—actually a standby position. Position 3, SINGLE, connects both outputs to the upper-track amplifier (for monaural reproduction of either stereo or monaural tapes.) Position 4, STEREO, connects left and right channels to upper and lower tracks respectively for stereo reproduction of in-line tapes. In addition to this switch, there is a playback volume control, two record volume controls—one for microphone and one for radio or phono inputs—and a PRESS TO RECORD button which furnishes plate supply to the oscillator and the record output stage and cuts out a resistor in the plate supply circuit to compensate for the additional voltage drop

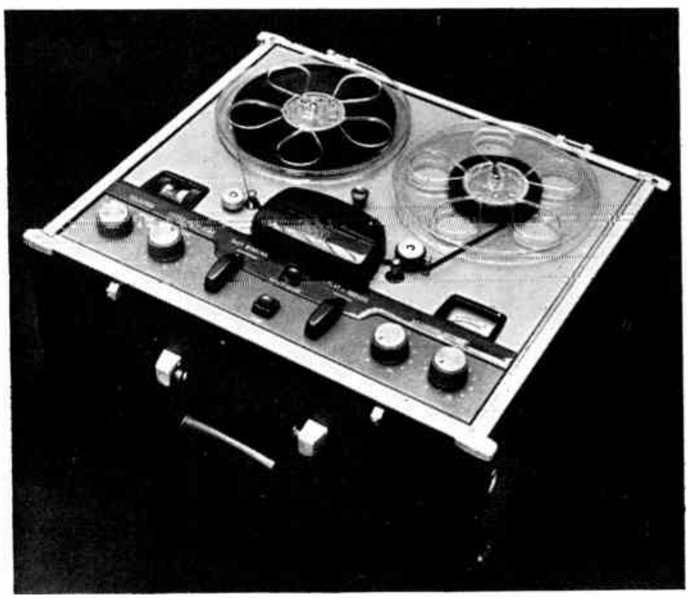


Fig. 1 (right). The Ampex A-122 magnetic tape recorder-reproducer—this particular model records and plays back monaurally, and plays stereo tapes.

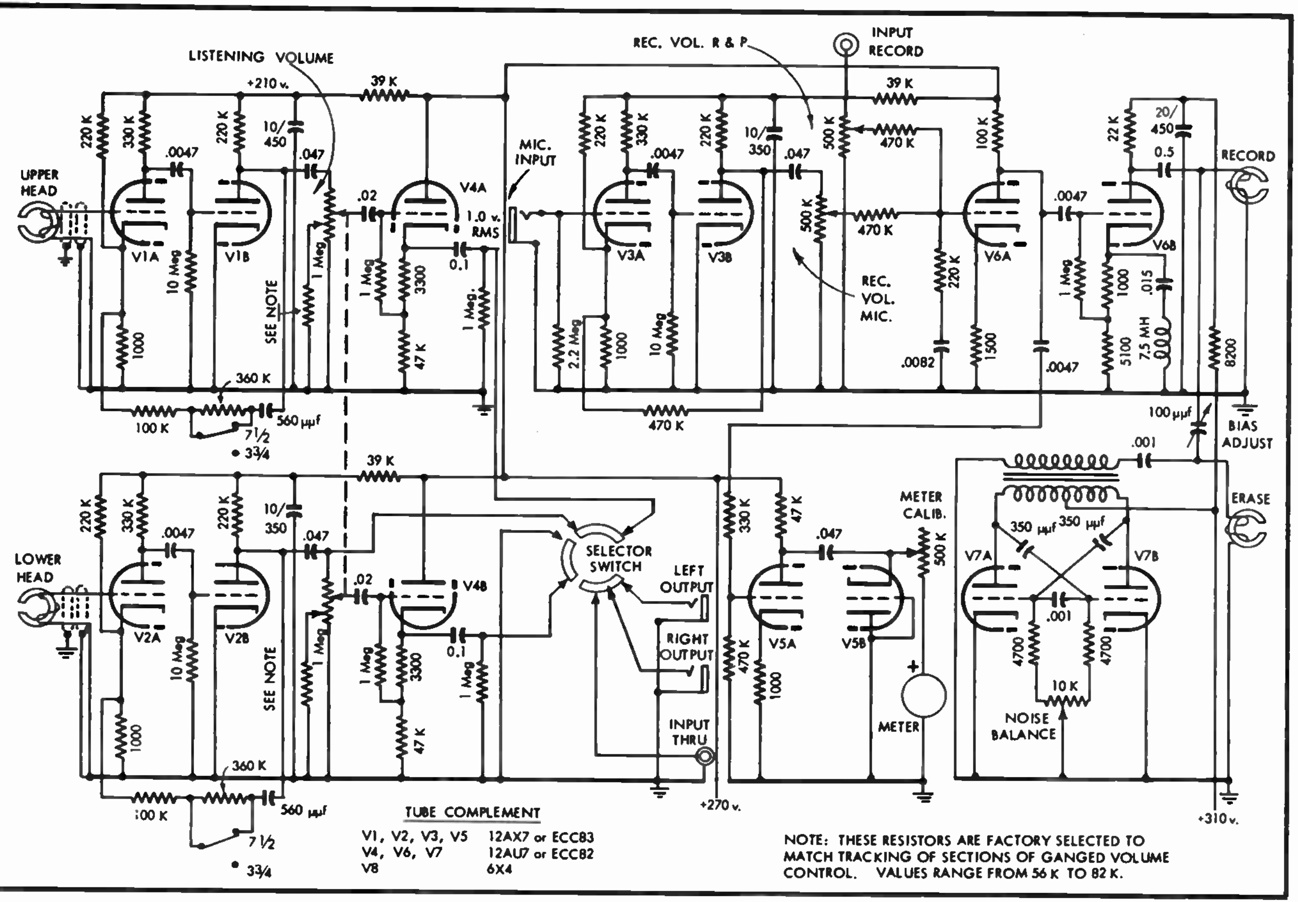


Fig. 2 Complete schematic of the A-122 recorder-reproducer, except for the power supply section.

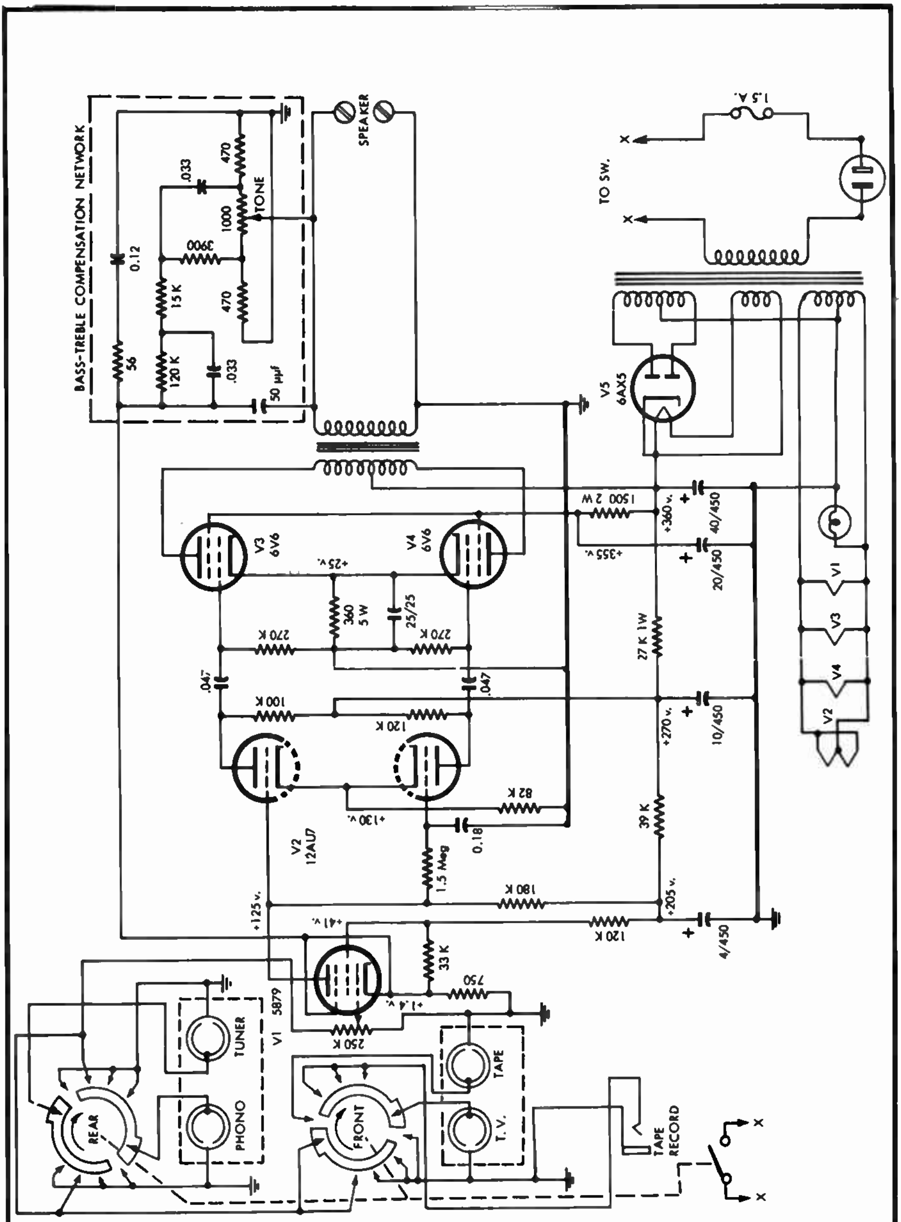
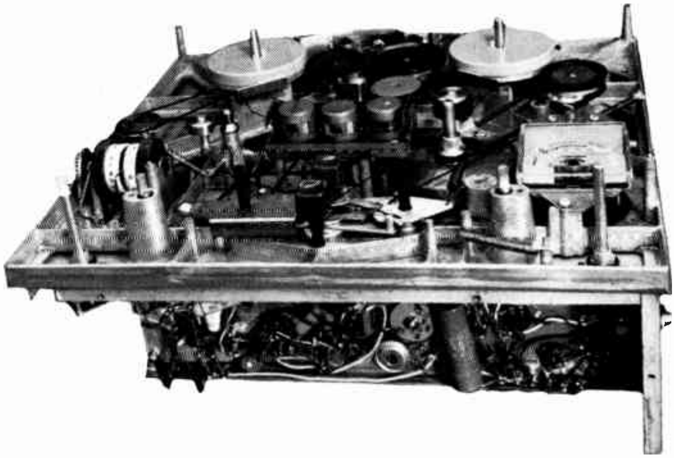


Fig. 4. Schematic of the A-692 amplifier-speaker. Note the compensation network and tone control in the feedback circuit.

Fig. 3. Neat simple construction characterizes the A-122. Note use of cast aluminum chassis and integral mounting of electronic chassis.



across the filter resistor due to the increased current drain. There are four mechanical controls—the PLAY OR RECORD knob, which starts the tape at the running speed; the FAST WINDING knob, which provides for fast forward or rewind; the STOP button, which stops the tape motion; and the speed selector, which is pulled up for 7½ ips, or pushed down for 3¾ ips. These controls are interlocked and each causes a number of functions to occur.

The drive motor is a four-pole shaded motor which is more dependent on voltage for consistency of pitch than the hysteresis type, but with the new method of holdback tension—a spring-loaded felt pad pressing the tape against a guide—the absolute speed is held within a specification of ⅓ of a half tone. Neither wow nor flutter is *audible* at all with this arrangement, and Ampex claims that the specification for the 122 is identical with that for the much more expensive Model 350. One crucial test is to record a tone on a strip of tape at the beginning of a reel, and then again at the end. Removing the two pieces of tape and splicing them together so they may be run through the machine successively will show the timing accuracy as a change of pitch as the splice passes the heads. We measured a pitch change of 10 cps on a 1000-cps tone, which is 1 per cent.

In-and-out IM distortion at the normal recording level—10 db below the point of 3 per cent harmonic distortion—measured 0.4 per cent. The response from 50 to 10,000 cps from a standard tape was within ± 1.5 db throughout, and recording a tone from an oscillator gave a range of ± 3 db from 30 to 13,000 cps, with 15,000 cps being down only 4.2 db from the 1000-cps level.

We have played a large number of stereo tapes on the A-122, listening very carefully, and from a purely subjective standpoint the performance is more than satisfactory. With the LISTENING VOLUME control at maximum, the hum is barely audible—not at all at the normal position at "7" on the dial. The head assembly—visible in *Fig.* 3, —is doubly shielded. Each head is encased in its own shield, and another shield (not in place in the figure) covers all three heads together. A "dress" cover of plastic gives a finished appearance to the machine.

A-692 Amplifier Speaker

As a companion piece to the A-122, the A-692 amplifier-speaker makes a complete system. This unit incorporates an Ampex-built 8-inch speaker mechanism in a housing designed for it, together with a 10-watt amplifier. The circuit of the amplifier is shown in *Fig.* 4, primarily to point up a feature that we believe to be good engineering—that of designing an amplifier to work with a given speaker and housing with correct equalization so that a "flat" signal fed into the amplifier appears as an acoustically "flat" output from the speaker. The means of obtaining this is to introduce a fixed amount of bass and treble boost—in this particular instance—to compensate for (1) the small enclosure, and (2) h.f. losses due to padding in the cabinet and natural rolloff of the speaker. Within its power capabilities, the combination serves as a very successful acoustic transducer, with essentially flat response from about 60 cps to over 11,000. For demonstration work and for complete portability, this unit is ideal; for stereo use two of them make a system that offers good quality and extreme compactness. The same system is employed in the furniture models as in the portables that we used for test, and the results seem to be about the same. For a fixed home installation we would prefer a further extension of bass response such as one would expect from conventional cabinets, but especially for stereo use the A-692 speaker-amplifiers do an excellent job—particularly when one realizes that both speaker and amplifier are housed in rather less space than most speakers alone.

While many tape recorders have been offered heretofore with the tape-transport mechanism separate from the electronic section, this can lead to trouble from hum pick-up because of the possibility of changing relationships between the playback head and the power transformer. In the A-122, the chassis is permanently attached to the transport chassis so that these relationships remain fixed, and once adjusted for minimum hum there is very little likelihood of a change taking place. The electronic chassis is L-shaped, and accommodates all tubes, transformers, and other elements of this section. This is undoubtedly a more economical method of construction, since there is no need for plugs and cables, nor is there any need for long leads from the amplifiers to the panel-mounted controls. All connections are made to a panel at the rear (except the microphone connection, which is a jack on the right end of the case), and all leads are detachable. Thus when the unit is removed from the case for servicing, there are no dangling leads.

The transport chassis itself is an aluminum casting, and is sufficiently rugged that one would expect alignment of bearings to remain constant for a long life. The entire unit is sturdy and gives the impression of being good for a long life in the field.

PENTRON TAPE EQUIPMENT

Using the "building block" form of construction, the current Pentron semi-professional line of tape equipment provides a great amount of flexibility to the experimentally inclined, and makes it possible for anyone to start out simply and later build up to as complete a tape recording system as he desires.

Starting with the TM-4 tape deck it is possible to add separate amplifier units so as to accommodate eight different requirements:

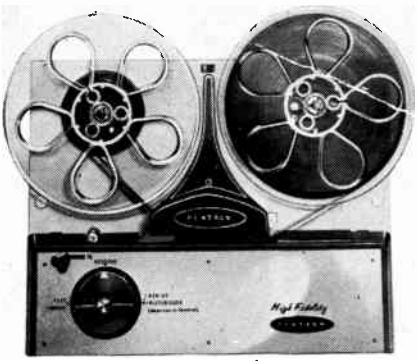


Fig. 4. Pentron TM-4 tape deck, one of the components of a complete system.

1. Monaural playback only
2. Monaural recording and playback
3. Staggered stereo play plus monaural recording and playback
4. Staggered stereo play only
5. Staggered stereo play/record plus monaural play/record
6. Stacked and staggered play plus monaural play/record
7. Stacked and staggered stereo play/-record plus monaural play/record
8. Stacked and staggered stereo play and monaural play.

Not all of these combinations require the TM-4 deck, as will be noted, but this model can be used for any one of the set-ups, and conversion can be made to step up the over-all flexibility. For these eight combinations, the following equipment will suffice:

1. TM-1 and CA-11
2. TM-1 and CA-13
3. TM-3, CA-11, and CA-13
4. TM-3, CA-15
5. TM-3, two CA-13
6. TM-4, CA-11, CA-13
7. TM-4, two CA-13
8. TM-4, CA-15

TM-4 Tape Deck

This deck, shown in *Fig.* 4, mounts vertically, horizontally, or at any angle, and records stereo or monaural when used with the required amplifier units. It operates at 7½ and 3¾ ips, and has single-knob control. A quarter turn to either right or left gives rewind or fast forward, respectively. Depress the knob and turn a quarter turn to the right and it records and plays. It is equipped with two heads—one a combination record/play head for a half track, and the other a stacked stereo head. For stereo recording—bulk erased or fresh tape is required.

The machine handles tape easily and smoothly, with reasonably light but effective braking. It mounts in a cutout 10-3/32 by 13 in., and requires a depth of 7 in. for clearance. Flutter and wow are claimed to be under 0.4 per cent at 7½ ips, and under 1 per cent at 3¾ ips. All heads have removable pole pieces so they may be changed readily after wearing. The unit rewinds 1200 feet of tape in a measured 98 seconds, with fast forward being somewhat faster—of the order of 80 sec.

Amplifier Units

The CA-11 preamplifier, shown in *Fig.* 5 is a 3-tube, self-powered tape playback amplifier which is equipped with a single gain control and a pilot light. It is 11-5/16 in. wide by 5 in. high and 8 in. deep (all amplifier units are the same size) and may be removed from its cage and mounted in a panel if desired. The equalization is modified NARTB, and the output is 1 volt from normal tape recording level.

Fig. 6. Pentrol CA-13 preamplifier—providing both recording and playback for one channel.

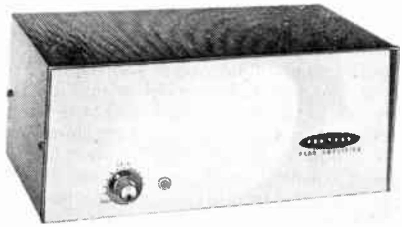


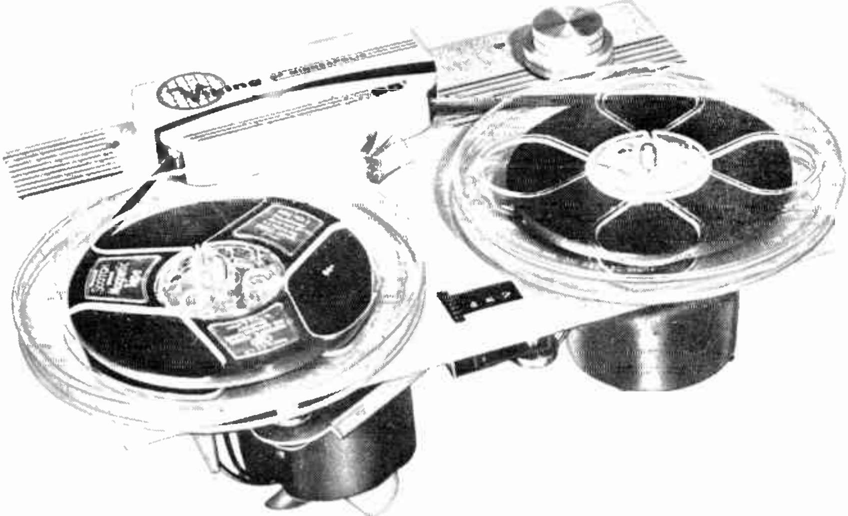
Fig. 5. Pentron CA-11 preamplifier—a single-channel playback unit.

The CA-13 preamplifier, shown in *Fig.* 6, is a record/play unit, with self-powered, and volume indicator and erase/bias oscillator. It will accommodate microphone or a high-level input such as radio tuner or phono preamp, and also serves for playback. An interlock switch prevents accidental erasing of desired material, and output level of the playback circuit is 1 volt at an impedance of 10,000 ohms.

The CA-15 preamplifier is used only for playback, and is essentially two CA-11's in the same cabinet. Separate level and equalization controls are provided for each channel, as well as a master gain control which changes gain simultaneously in both channels. This unit has five control knobs.

One other unit, the CA-14, completes the lineup. This is a microphone or phono mixer with four channels—two at high gain and two at either high or low gain as desired. This unit has a gain of only 8 db in the microphone channels, but it permits the use of as many as four mikes to feed into a single record amplifier with suitable mixing facility and no loss in over-all gain.

Taken together or in parts, this is a flexible line of equipment suitable for the experimenter who (1) needs the various services available, or (2) wishes to build up to a complete system with a step-by-step approach. The convenience of mounting provides for a wide latitude in housing the units, or they may be used in their separate cases and plugged together as required.



VIKING MODEL 85 TAPE DECK and RP-62 RECORD-PLAY AMPLIFIER

Practically anyone with a high fidelity system will admit that one of the important elements is the tape recorder, and when the tape equipment is operating properly the enjoyment of the entire system is increased immensely. One of the difficulties of installing tape equipment in a home system has been the lack of availability of good machines which were adaptable to building in except in the very-high-price bracket. Furthermore, most of the medium-priced units were complete recorders, with portable case, built in loudspeakers, and 2- to 4-watt "power" amplifiers. As far as the audiofan is concerned, all this is superfluous, since he wouldn't be caught dead with some of the speakers which are built into the portable cases, and he already has amplifiers which are better in both the quality and power departments.

The current Viking models, from the 75 to the 95, fit the home requirement admirably. The model tested was the Model 85, with a sub-designation "RQ," the "R" signifying a recording model, and the "Q" indicating that it would accommodate the quarter-track system. Practically any arrangement of heads may be installed on the Viking 85 to provide a variety of services, and the user would do well to check exactly what he wants when ordering a machine. The unit will accommodate up to five separate heads, in any order, and the heads available include: half-track erase, in-line half-track erase, half-track record/playback, in-line half-track record/playback, and in-line quarter-track record/playback.

The deck employs two motors, both four-pole types. The capstan is driven by a belt from the rubber floated motor platform, and a 1½-lb flywheel holds flutter and wow to less than 0.25 per cent. All controls are on the front panel, with fast forward and rewind actuated by the outer knob shown at the upper right of *Fig.* 4, while the bar knob selects the record/playback mode, or the CUE mode in which the brakes are released but the pressure pads hold the tape against the heads so that the reels may be turned backward or forward for editing and cueing. The two knobs are interlocked so as to prevent improper operation which might result in tape spillage or breakage. Two additional controls appear on the front panel—the speed control, just below the head housing, which sets the mechanism for 3¾ or 7½ ips, and the head-shift adjustment, which positions the heads for either half- or quarter-track operation. At the bottom of the panel is the digital counter, which aids in locating any portion of the tape that has been catalogued in advance.

As for handling, the tape deck is extremely smooth and convenient. We could find no normal operating procedure which either spilled or broke the tape (there are some *abnormal* operating procedures which will break tape, such as making a quick switch from fast forward to record, but that will happen with any machine). Tape tensions are moderate, brake action effective and consistent.

RP-62 Record/Playback Amplifier

While several types of amplifiers are available for use with the Viking decks, the RP-62 is one of the basic tvpes needed for recording and playback of a single-channel tape. For stereo recording and playback, two such units would be required—the second unit's bias oscillator being synchronized to the first so that no beat note will result from differing bias frequencies.

The amplifier provides for inputs from a tuner or other high-level source, or from a microphone, and from the record head on the tape deck. Outputs to the erase head and to the record head are provided, as well as an additional output to the amplifier for playback. The entire unit employs a 12AX7 and a 12AU7 in the amplifier section, a 12AV7 or a 12AU7A for the oscillator, a 6E5 as a recording level indicator, and a 6X4 as a rectifier. Rack-mounting models are available with a VU meter instead of the indicator tube, and a single preamp for playback only is also usable.

At first observation, we were inclined to wonder why the playback equalization was adjustable from the front panel of the amplifier unit, but after using the system for some time the advantages became obvious. The control permits a variation of response at 10,000 cps from 5 db below to 5 db above the standard NARTB curve, which makes it possible to accommodate tapes made on practically any machine. With a standard tape, however, one can easily determine the correct setting and for tapes made on the user's machine, he would be assured of uniformity in playback. In accordance with NARTB standards, the high-frequency boost is applied in the record amplifier, and the low-frequency compensation comes from the playback operation, so with the standard recording curve and an adjustable playback curve, consistent quality will result, and excellent results will be had from recorded tapes. For the home user of tape, the Viking system seems to fill a variety of needs in a fairly simple manner, and as the user's needs change, he may build up to them gradually.

BLONDER-TONGUE "AUDIO BATON"

New amplifiers, new tuners, new preamps, and new speakers are constantly coming on the market, but this is a unique device which has functions not duplicated by any other device commonly available. To be sure, there are octave and half-octave filters, but all of them we have seen so far are of laboratory quality, and are much too sharp in the cutoff region to serve for any other purpose than making measurements.

The Audio Baton, however, while *similar* to an octave filter, is rather better described as a "super-duper tone control." Pictured in *Fig.* 3, it consists of seven separate amplifiers each tuned to a specific frequency, together with another stage which is followed by a high-pass filter in parallel with a low-pass filter, all with their outputs fed to separate level controls, nine in all. These controls are accessible from the front panel as knurled knobs, mounted horizontally below the slots located at each octave point under a reproduction of a piano keyboard. Through the slots may be seen an illuminated white column, on which a red spiral is marked; as the knob is turned, the red stripe serves as an indicator to show the amount of boost or cut that is applied at that point in the spectrum, in effect drawing a frequency response curve on the simulated scale background. Each control provides a boost of 13 db at its particular point in the spectrum, with a 13-db cut throughout the midrange, a 11-db cut at the 80- and 5120-cps points, and a 6-db cut at 40 and 10,240 cps. The controls are designated with their operating frequency, and appear at 40, 80, 160, 320, 640, 1280, 2560, 5120, and 10,240 cps.

Technically, the circuit arrangement is as shown in *Fig.* 4. An input level switch may be set for two ranges—0.1 to 0.6 volts,

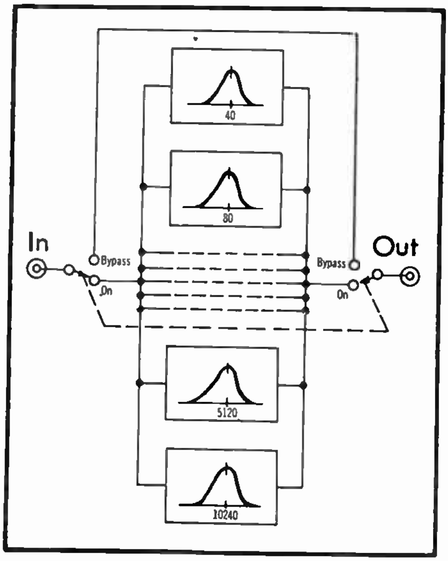
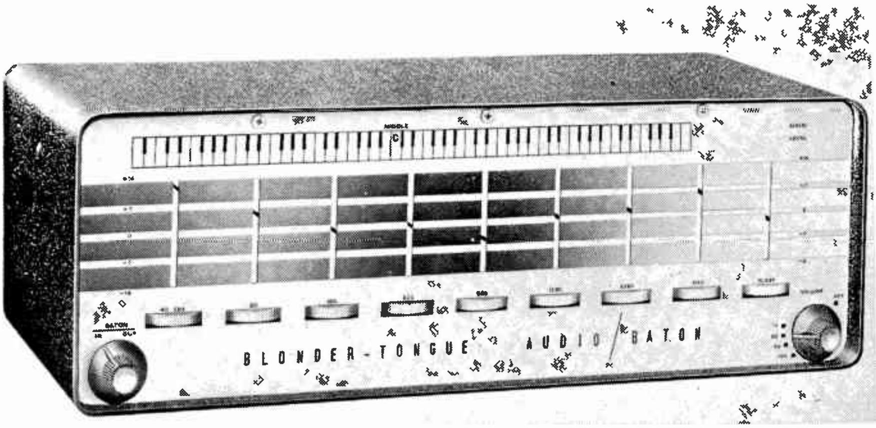


Fig. 4. Block diagram of the Audio Baton circuitry.

or 0.6 to 1.5 volts. The insertion loss of the complete instrument is zero when all controls are centered. The IN-OUT switch bypasses the frequency-correcting circuits and provides flat response (± 0.5 db)



through the two feedback-controlled output stages to an output level control. With the switch in the IN position, the equalizer section is in service, and with all controls set in the "0" position, response is essentially flat from 20 to 20,000 cps. Actually the response droops about 1.5 db between each of the "peaks" since the over-all response is the sum of the responses of the nine separate circuits, seven having responses which are essentially triangles with slopes of about 8 db per octave on either side of the center frequency, with the two filters—at the ends of the spectrum—having slopes of around 6 db per octave.

Performance

Distortion of the instrument in the OUT position of the switch was measured at several points in the spectrum, and was constant at 0.2 per cent. In the IN position, distortion was somewhat higher, but still less than 2 per cent anywhere in the spectrum. Hum and noise measured 64 db below the rated output of 1.5 volts. Measuring the response with all controls except 1280 cps at the − 14 db position, and with the 1280-cps control at the + 14 db position gave an output 30-db down at 100 cps and at 18,000 cps, with practically straight sides to the curve, reaching the reference level at 1280 cps. Obviously there would be some effect on the adjacent octave controls with a slope of only 8 db per octave, but in actual listening it is unlikely that one would set adjacent controls at widely varying positions.

Applications

The Audio Baton has a number of interesting applications other than the more obvious one of serving as an adjunct to a typical high fidelity music system. As a matter of fact, it is likely that the instrument will find its greatest popularity among recording studios—being considerably less expensive than equivalent devices produced for the professional field. We did some experimenting with dubbing tapes of unsatisfactory quality such as might be encountered from the garden-variety of camera-store tape recorder, and we would consider the device indispensable to anyone interested in dubbing from old records or from tapes which are not quite satisfactory. There is no doubt that the Audio Baton can do an excellent job of correcting frequency response of the less serious types likely to be encountered in audio equipment. Rolloff of either lows or highs due to poor microphones or narrow-range amplifiers or recorders can be corrected easily, and response peaks can be smoothed out as desired.

For ordinary listening, it is possible to increase or decrease the presence effect, moving an instrument or a voice "out in front" or pushing it back at will. Telephone effects are readily obtained, as are a number of other special effects, and clarity of speech can be obtained by removing the chestiness occasioned by a bass control which extends too far up the scale. Similarly, screechiness can be eliminated completely.

One of the uses of rather great importance to PA system operators is in the elimination of acoustic feedback or "howl." By reducing the response at the howl frequency, it is possible to increase output level as much as 10 to 12 db without causing an appreciable deterioration of the sound quality as noticed by the human ear. As one howl frequency is corrected, another will crop out as the volume is increased, and with a few variations in setting of the controls the over-all output level can be increased very effectively.

With the Audio Baton it was easily possible to simulate the effect described by Staffen in the article commencing on page 21 of this issue. The advantage of the increase in very-low bass is quite readily apparent.

Another interesting application is in the direct comparison of two pieces of equipment—particularly loudspeakers in A-B testing. Switching from A to B while at the same time switching the Audio Baton in or out allows the listener to adjust the Baton so that both speakers sound as exactly alike as possible. The difference between them is then instantly observable on the scale of the device.

For those applications where the Audio Baton is continuously in circuit, there is one feature that would be considered desirable, even though unexpected. The nine slots through which the illuminated dial cylinders are seen are quite bright with a full 6.3 volts applied to the pilot lights. A three-position slide switch on the rear apron makes it possible to insert resistance into the pilot-light circuit in two values, giving three degrees of illumination. We would prefer larger and more legible panel designations for frequency and degree of boost or cut, but after thorough familiarization it is likely that this deficiency would not be noticed. In general, we would recommend the Audio Baton very strongly to anyone who does much recording or dubbing of tapes or discs, and for any PA application where acoustic feedback might be encountered. While it is relatively simple to correct such troubles in permanent installations, there are many times when a PA system is set up for a one-night stand and it is not practicable from the standpoint of time and cost to effect a permanent cure. The Audio Baton could well be indispensable to the portable PA system operator.

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