PRACTICAL TRANSISTORISED NOVELTIES FOR HI-FI ENTHUSIASTS

BY

B.B. BABANI

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Although every care is taken with the preparation of this book the publishers will not be responsible for any errors that might occur

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SPECIAL NOTE FOR AMERICAN READERS

In order to help readers of this book, a table of transistor equivalents showing American types that may be used in place of the British types shown in the circuits in this book.

BRITISH	U.S.A. TYPES
BC109	RS276-2009 Archer (Radio Shack & Allied)
BC109	GE20 General Electric
BC109	TR21 International Rectifier
BC109	PTC132 Mallory
BC109	HEP 53 Motorola
BC109	SK3020 R.C.A.
BC109	RT100 Sprague
BC109	ECG123 Sylvania
BC109	2N3565 Any U.S.A. Manufacturer
2N4360	Available from any U.S.A. Manufacturer

The compiler of this book wishes to express his most sincere thanks to 'ELECTRONICS AUSTRALIA' the leading radio, T.V. and electronics magazine published in that continent, and to Syndication International Ltd., their agents, for permission to use much of the material in this book which appeared originally as articles in that magazine.

QUADROPHONY

ADD TWO EXTRA CHANNELS TO YOUR STEREO SYSTEM

Here's a simple little decoder unit that will allow you to sample some of the pleasures of four-channel reproduction without having to dig too deeply into your wallet. It will provide acceptable decoding of matrixed records currently on the market but, more importantly right now, will add a whole new dimension to your existing record library. Get it going and our tip is that you won't want to part with it!

A lot has been said about four-channel reproduction and it should be sufficient to recapitilate here only briefly to provide a background to the present project.

It seems almost certain that four-channel reproduction will feature very large in the future of high fidelity reproduction and that it will be introduced to the mass market by way of the disc. Tape isn't strong enough yet to effect a revolution on its own but tape will follow suit and, in due course, the tape/disc tussle will continue on the four-channel arena.

Right now, the problem of the four-channel disc is one of standards. On one side a powerful group headed by RCA in America and JVC/Nivico in Japan is backing what they call the "discrete" system – in which some of the essential information is imposed on a high frequency carrier modulated onto the groove walls along with more conventional audio modulation. The method requires a special pickup and a special demodulator but it produces a high order of separation between channels.

A powerful rival group headed by CBS in America and Sony in Japan is backing the "matrix" method – technically less ambitious but more convenient and possibly adequate to the market needs. However, the matrix method is itself segmented by rival companies who prefer their own particular approach and their own particular set of patents.

While the individual big companies are pushing the four-channel concept along with their own favoured system, hifi interests outside the committed companies are being cautious, in case they should end up backing the wrong system. In fact, many hifi merchants have been tending to sell down the whole idea on the basis: "stereo you can be sure about; four-channel is anybody's guess!"

In the four-channel amplifiers currently being offered on the market, the most common approach is to provide a switching facility which selects one or two in-built decide matrices and "discrete" input. In this last position, inputs to the four channels, are simply brought out for possible connection to an external demodulator or decoder or a four-channel tape system. Few equipment manufacturers have tried to build in full facilities for all the possible systems.

Given enough patience, it would doubtless be possible for an enterprising enthusiast to sound out the details of the discrete and the contending matrix systems, do the sums and evolve a design for an all-purpose switchable decode unit. It would involve a great deal of time, effort and expense and may be rather hard to justify.

But, equally, many enthusiasts will not see this as a reason to ignore the new technology altogether. It is much too intriguing!

Our own experience with "surround" (if not four-channel) sound is that it can be very pleasant indeed and, after having lived with it for a while, ordinary two-channel stereo sounds rather tame. Even for classical music, where one is likely to prefer the orchestra plainly out front, that sniff of ambiance from the rear can add significantly to the aural interest.

Undoubtedly the cheapest and the simplest way to achieve a surround effect is by the Hafler/Dynaco method.

However, it does lack flexibility. Problems multiply rapidly if provision is sought to control level and tonal balance or to adapt the idea so that it can perform matrix decoding as well as simulation from two-channel stereo recordings. Over and above the circuitry and the components involved, there is the problem of avoiding loss of power in the main loudspeakers and/or adverse loading on the amplifier channels.

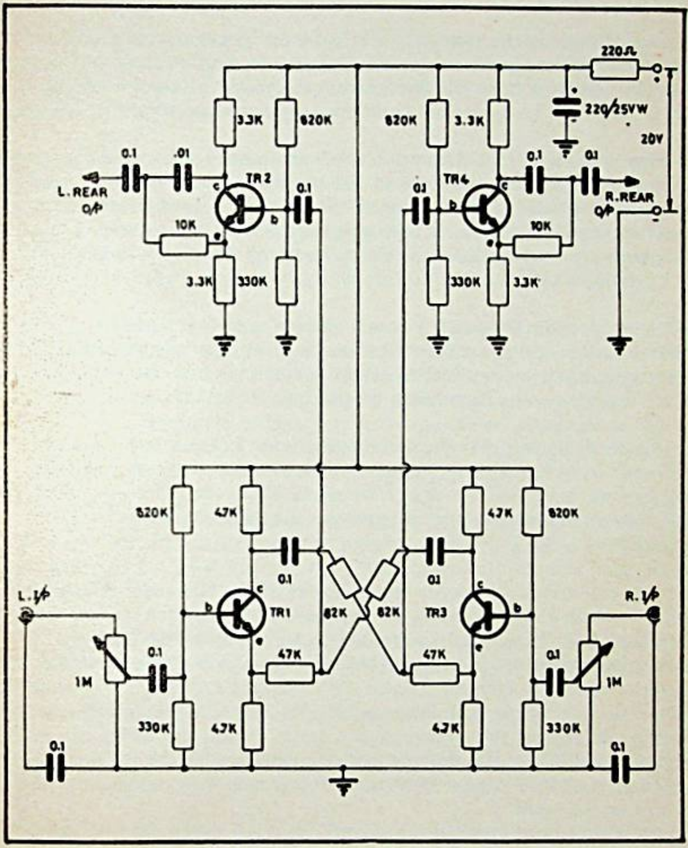
Prompted by these considerations, we recently took a closer look at the various matrix decoding arrangements, along with networks currently being used to simulate four channel sound from conventional stereo records.

At one extreme we could have ended up with anything from 12 to 20 transistors, or multiple ICs, with a function switch and a circuit board full of bits. We may, in fact, go further along these lines if such a course seems warranted by the demand and a build-up of interest on four-channel records.

At the other extreme, we found ourselves with a very simple alternative using only four transistors, a modest number of wiring components and no switching. It is along very similar lines to simplified circuitry found in some recent Japanese amplifiers.

As planned, the unit is intended to operate in conjunction with a second stereo amplifier driving a pair of loudspeakers in the other corners of the listening room. It derives its input signal from the main stereo amplifier, most commonly from the loudspeaker output circuit.

When the main amplifier is reproducing the signal from a conventional stereo record, the add-on system provides extra signals for the rear channels, different in content and phase from each other and from either of the



4 CHANNEL STEREO DECODER

Involving only four transistors and a handful of small components, the new adaptor unit will synthesise rear channel signals either from existing stereo records or from the new quadraphonic pressings.

front channels.

When the main amplifier is reproducing the signal from a matrixed four-channel record, the additional circuitry performs partial decoding, providing a quite convincing surround sound.

Because of its decoding function, it offers more to begin with than the simple loudspeaker network mentioned earlier. It does not load the main amplifier and at all times the level, tonal quality and balance of the rear loudspeakers can be controlled from the supplementary amplifier.

Looking at the circuit of the decoder, left and right signals derived from the main amplifier are impressed across two tab potentiometers, which can be set to ensure a balanced and acceptable level of signal to the base of the respective input transistors. Blocking capacitors have been specified in series with the earth lead, in case the common of the main loudspeakers is not at chassis earth.

The input transistors for each channel operate as phase splitters, such that in-phase and reverse-phase versions of the left and right signals appear at the respective emitters and collectors. These provide the four sources from which the rear channel signals can be matrixed (or mixed).

The signal for the left rear channel is constituted from an in-phase sample of the left front signal (from the input transistor emitter) and a smaller, reverse-phase sample of the right front signal (from the other input transistor collector). A complementary mix provides the signal for the right rear channel.

The proportions are set mainly by the choice of the 82k and 47k resistors which establish a ratio between the in-phase and reverse-phase components of 1.0 and 0.55. These matrixed resultants are impressed on the bases of two further transistors from which the rear output signals are derived.

Where a decode matrix is an integral part of a four-channel amplifier, it is usual to feed it from the preamp and to derive from it signals for all four of the power amplifiers. It is then possible to predetermine the relative phases and levels of all four signals. Deliberate frontal cross-talk can also be introduced if desired.

Proportioning of the signals is one of the large areas of contention surrounding the matrix method and is one reason why, at present, a single universal decode matrix is not really practical.

Where the decode matrix is external to the main amplifier, as with the unit described here, the matrix can only set the differential phase and amplitude of the two rear channels. Their level relative to the front loudspeakers is a function of the volume setting of the second amplifier. (In fact, there is nothing unique about this because, after the matrix designer has done all his sums, most four channel amplifiers provide left-right and front-back balance controls, to give the operator the last word anyway!)

The obvious point of note about a purely external matrix is that it cannot introduce left-right frontal cross-talk into the main system. If this is desired, it may have to be introduced separately into the main amplifier by modifying the internal circuitry.

Two possible suggestions come to mind. Many older amplifiers have a stereo reverse position on their mode switch. This could conceivably be rewired to obviate the reverse function and to introduce a selected value of resistor between the left and right preamplifier channels. Alternatively, the mono position could be modified, to eliminate the direct inter-channel link and to substitute a resistor which would give only partial coupling.

Whether or not such a link is deemed necessary can be the subject of individual observation and experiment. If the aim were purely to decode four-channel recordings, it would be provided. In a general-purpose matrix which will be used most of the time for creating a new sound field from conventional stereo recordings, it may be considered unnecessary, even undesirable. However, if the mode switch of the main amplifier can be modified easily, you can have it both ways!

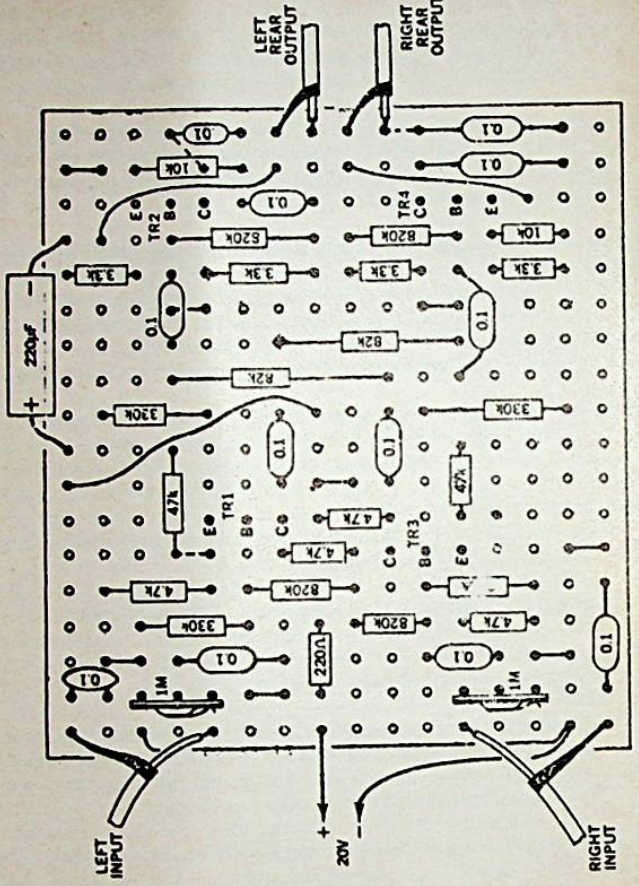
And here a word about channel separation. With two-channel stereo, the ability to achieve a large degree of separation has been considered vital. It can become a fetish to the point where enthusiasts seem almost to listen to the separation rather than the program!

With a sound field that encompasses the listening position, the same preoccupation with source separation is not necessarily justified. Many hold to the view (admittedly a commercially convenient one) that sharp definition of the sound sources is neither necessary nor pleasant.

This is one of the things about which you will be able to make up your own mind.

But back to the simple decoder: The output for each rear channel is taken from a reactive divider strung between the collector and emitter of the respective output transistors. This does not affect the frequency response but it does produce a rotation of phase with frequency.

At low frequencies, where the reactance of the capacitors is high, the signal is derived primarily from the emitters. At high frequencies, where the reactance of the capacitors is low, it is derived primarily from the collectors which are, of course, 180 degrees removed in phase from the emitters. At



Here is the component layout on the Veroboard version of the 4 Channel Stereo Decoder unit intermediate frequencies the output signal phase will fall somewhere between the two.

The two capacitors in the circuit have deliberately been chosen to give different orders of phase rotation between the two rear output signals. Below 100Hz the two output networks deliver a signal which is substantially that at the respective emitters. Above 2.5kHz, the signals are substantially those at the two collectors. Over most of the intermediate range, where each of the output signals is undergoing phase transition, the difference between them approximates 90 degrees.

The majority of two-channel records are so recorded that the bass content of the two channels is in phase. This ensures good foundational bass and does not markedly deteriorate separation because listeners are not acutely aware of the direction from which pure low frequency energy comes. (Overtones yes, but not the fundamental).

Where the rear loudspeakers are fed with a difference signal only, as with the simplest Hafler type network, the bass content is almost completely cancelled. As a result, reproduction from the rear channels is "thin", appropriate for ambience but unacceptable at too high a listening level.

By contrast, the matrix featured in this article permits only partial cancellation of signals common to both channels. It therefore retains a substantial bass component in the rear channels, derived virtually from the respective emitters and not subject to differential phase shift. With normal bass content, the sound from the rear loudspeakers can be tonally balanced and entirely acceptable in its own right. Furthermore, by ensuring that the rear cones operate in phase with the front cones (ie tend to compress air in the listening room simultaneously) they can reinforce the bass response overall.

If something were not done about it, this additive effect would create a problem at middle frequencies. A strong centre-front signal (virtual mono) would produce strong in-phase components from the rear channels and create a competing centre-rear image. This is where the 90-degree phase differential plays its part. Without changing the tonal balance in the rear loudspeakers, the relative phase rotation tends to defeat the tendency merely to image the front channels.

Above 2.5kHz the rear channels again operate in phase on common signals but at these progressively shorter wavelengths, relative phase and amplitude at any given listening position become much more random anyway. Some designers have used more complex networks which spreads the phase differential over a wider range but, for our present purpose we preferred to keep to the simple circuitry.

The decoder is intended to operate with an effective rail voltage of 18-20, at a current drain of about 5 milliamps. For preliminary evaluation it could be operated from two series-connected 9V batteries but this could hardly be regarded as a long-term arrangement.

In fact, it should be possible to obtain the requisite supply from most amplifiers, without undue difficulty. Where the supply rail in the amplifier is only around 20V, the decoupling resistor should be kept small in value, the suggested figure being 220 ohms. With high rail voltages, the resistor must be increased before the unit is connected to a value such that it will reduce the voltage at the decoder to not more than 20.

With amplifiers having a negative supply rail, the simplest course would probably be to substitute general purpose PNP transistors and reverse the polarity of the supply and the electrolytic decoupling capacitor.

Because of the input potentiometers, the decoder can cope with any signal level likely to appear across a loudspeaker voice coil. Overload level at the actual input base is about 3V RMS nad it is therefore advisable to keep the input signal level well below this.

The available output should, however, be ample to feed into the "auxiliary" channel of any ordinary system providing the volume control is operated about nearly full on. At these signal levels hum is not likely to be a problem and the exact location of the decoder is not very critical.

Some of our tests were done with the decoder mounted inside a 2 x 3-watt stereo amplifier. It was simply hung under the front lip of the chassis and powered from the supply rail. The wiring was modified so that signal from the main loudspeakers was fed in through the auxiliary input to the decode unit, and thence to the auxiliary contacts on the mode switch.

Wired in this way, the supplementary amplifier becomes a self-contained quadrophonic adaptor. It needs merely to be connected to the existing system and to an extra pair of loudspeakers and set up for optimum listening level. Thereafter normal control can be via the main amplifier.

While almost any existing stereo amplifier can be pressed into service, one of our staff went one better. Not having a spare amplifier and loudspeakers available, he arranged for his stereo tape recorder to provide the rear channels, using it simply as an amplifier/loudspeaker system.

Incidentally, this same member of our staff, pointed out that many readers would be likely to have other transistors on hand which could be used in this circuit – notably BC108, BC208, TT108 or 2N3565. While the last four transistors are not equivalent to BC109s, the circuit configuration should permit their use.

Fairly obviously, the Wired Veroboard could be housed in a small metal box, if so desired, rather than being housed inside the main or supplementary amplifiers.

A relatively small amplifier and modest loudspeakers will provide a very worthwhile surround effect, provided the listening room is not too large and the bass is not boosted beyond the amplifier's capabilities. It should be remembered that any bass boost applied in the main amplifier will also be effective for the rear channels; in fact, bass boost in the supplementary amplifier could be regarded as something of a luxury.

However, while the atmosphere of surround sound can be created with modest equipment, there is everything to be said for upgrading the rear channel equipment to the limit of your resources. The nearer your listening position is to the centre of the room, the greater will be the inclination to have all channels operating at the same level.

Perhaps we should hasten to add that this is the inclination for popular music, dinner music or mood music. You may even stop worrying about where you sit in the room because the room is simply and pleasantly full of sound.

For a concert-hall atmosphere, the levels will need to be adjusted differently. That's unless you get to like it the other way.

Happy listening!

YOU WILL NEED THESE COMPONENTS

Veroboard or similar copper backed matrix board, 3¾ x 3⅜ inches. Hole spacing 0.2 inches.
Shielded stereo cable, as necessary.
2 1-inch threaded spacers (⅛ Whitworth thread)
4 countersunk head ⅛ whitworth screws, ¼ inch long.
Hookup wire for supply leads.

SEMICONDUCTORS
4 BC109, BC209 or similar transistors.

CAPACITORS
1 220uF 25V electrolytic.
11 0.1uF, 1 0.01uF miniature polyester capacitors.

RESISTORS

4 820k
4 330k
2 82k
2 47k
2 10k
4 4.7k
4 3.3k
1 220 ohm

VARIABLE RESISTORS
2 miniature preset pots, 1M

Note: Resistor wattage ratings and capacitor voltage ratings are those used for our prototype. Components with higher ratings may generally be used providing they are physically compatible. Components with lower ratings may also be used in some cases, providing the ratings are not exceeded.

MEASURING AUDIO POWER OUTPUT

Power output measurement is one of the more important tests we should apply to any new amplifier. It is relatively simple, yet can yield a good deal of valuable information about the amplifier's performance. This article discusses the advantages of the measurement, the simple precautions needed to perform it correctly and the British and American standards laid down for testing commercial amplifiers.

One of the most important tests which can be carried out on a new amplifier is that for power output. It will help to establish more certainly than anything else whether the amplifier as a whole, and particularly the output stage, is operating as intended. Failure to reach the specified figure is a clear indication that operating conditions need checking or modifying before proceeding with any other tests.

Yet it is surprising how seldom this test is made or, if it is made, how seldom it achieves a useful degree of accuracy due to the manner in which it is conducted.

This is all the more so since it is a relatively simple test, capable of good accuracy and requiring little in the way of equipment not normally accessible to many audio enthusiasts.

In an amplifier which uses an output transformer there are two ways in which this test may be approached. One is to measure power developed in the primary circuit of the output transformer and the other the power available in the secondary and which is available at the voice coil of the loudspeaker.

Where the circuit does not employ an output transformer, the power at the output socket and available for the voice coil of the loudspeaker is, fairly obviously, the only measurement that has to be made.

Regardless of the approach chosen, both involve the same broad principles, namely feeding the audio power into a load resistor of the appropriate value measuring the audio volts developed across it at the overload point and, from the figures, calculating the audio power output.

Assuming the use of an output transformer, the measurement of primary power is useful mainly in development work, as when it might be desired to establish the correct load for a set of unpublished operating conditions. Its main advantage is that it is largely dependent of losses or variables in the output transformer which might otherwise confuse the issue.

Secondary power is a more useful figure when we wish to know just how many watts are available to drive the loudspeaker. It is less decisive as a measure of the output stage operating conditions. However, if a reasonable estimate can be made of the output transformer losses it nevertheless provides a useful guide to the amplifier's performance.

It is most useful where, a design having been worked out and accepted, it is desired to establish that any individual amplier made to it does in fact approach the output claimed and is therefore unlikely to contain any major faults.

In the case of a transformerless output stage there are no consequent losses to cloud the issue, so that a power output measurement provides, at one and the same time, an indication that the output stage is functioning as intended, and a figure which can be quoted as the output power available to drive the loudspeaker or other load.

Inasmuch as the main purpose of an amplifier is to deliver a specified amount of power to an appropriate load, one would expect that this would be the natural measurement an enthusiast would look forward to making when he had tightened the last nut and bolt and deposited the last blob of solder. Yet so many seem content to simply give the system a listening test, and, providing it sound "good and loud," accept it as OK.

And even when results are disappointing, and they seek someone's advice, it is seldom that they can quote precise figures to support their claim that it is not delivering the power it should. They are usually content to mentally compare it with some other amplifier, rated at so many watts, then pluck a figure out of the air to suit their own unit, depending on whether it sounds louder, the same, or weaker.

In the case of high-power PA amplifiers and guitar amplifiers, which may be rated as high as 100W, or even some domestic equipment which may run as high as 30W, measurement – rather than subjective reaction – is even more important. It is virtually impossible to run such amplifiers "full bore" into a loud speaker load on the bench, or to assess anything worthwhile from such a test should it be attempted.

It should also be appreciated that other vital amplifier characteristics (sensitivity is one example) are normally quoted on the basis of maximum power output, so we must be able to measure power output before these other characteristics can be assessed. On the other hand if, when a output measurement is made, we can also measure (for example) the level of signal being fed into the amplifier, the test will yield both power output and sensitivity figures.

Having this made a case for this test, let us take a closer look at what is required to perform.it. First, the load resistor, or "dummy load" as

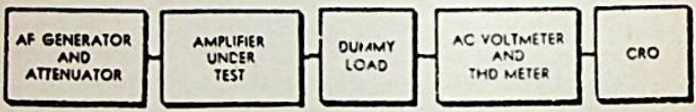


Figure 1.

Block diagram showing the set-up for measuring "Continuous Power Output" or "Rated Power Output".

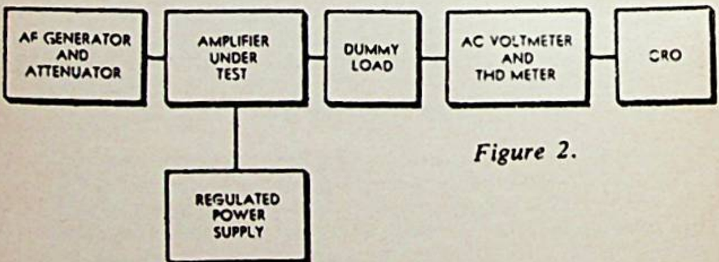


Figure 2.

An early method used to measure "Music Power". It had many limitations and is seldom used.

it is usually called. This is to replace the loudspeaker, which cannot be used for several reasons, the main one being that it simply is not accurate enough, in terms of impedance, for the purpose.

Impedance values quoted for speaker voice coils are purely nominal and are quoted for one frequency only. The actual value likely to be presented in a typical test set-up could vary by two to one from the nominal.

In its place we substitute a resistor, Ideally, this should be a non-inductive, close-tolerance unit, capable of dissipating the anticipated power without damaging heat rise or change of value. In practice, we can tolerate the small amount of inductance inherent in a simple wire-wound unit, and a resistance tolerance of 5% would be adequate in most cases.

In regard to power handling ability, it should be appreciated that resistors are normally rated on a "free air" basis and should be down rated if they are enclosed in any way. Except for intermittent use a good rule-of-thumb is to half the wattage rating when the resistors are to be enclosed in a metal instrument case, provided with reasonable ventilation.

The value of the load resistor will depend entirely on the particular job to be done. It could range from (typically) 10,000 ohms for a primary measurement of a valve type amplifier, down to (typically) 8 ohms for a voice coil measurement.

In practice, it is most convenient to make up a load box containing a representative group of heavy-duty resistors, complete with the necessary switching, terminals, etc., to make for greatest convenience.

Where the load is to be substituted directly for the voice coil the arrangement is so straightforward as to warrant little comment. The primary measurement is, however, somewhat different. Because there is both AC and DC present in the primary circuit, we must provide circuits for each. Simply substituting a resistor for the transformer primary would result in an intolerable voltage drop and failure of the output stage to function correctly.

The transformer is therefore left in circuit, but with all secondary load removed. It thus acts as a high impedance choke, and has little effect on the load as seen by the output stage. On the other hand it continues to function as a low resistance path for the DC supply. The dummy load–equal in value to the required primary impedance–is then connected in parallel with the primary winding, where it is seen by the output stage exactly as if it had been reflected from the low impedance secondary circuit.

Next, the voltmeter. This needs to be a good quality AC instrument, with a suitable selection of ranges, In addition, a DC blocking capacitor will be needed whenever measurements are to be made on the primary side of a transformer. Even if the meter is connected directly across a transformer primary, there may be a small DC voltage present, due to the resistance of the primary winding. There is no such problem when measurements are made in secondary circuit and the normal AC ranges may then be used.

For measurements at voice coil impedance, the sensitivity of the meter is not particularly critical. Anything above about 100 ohms per volt should be perfectly satisfactory. For higher impedance measurements, such as those encountered in the primary circuit of valve amplifiers, the meter impedance should be at least 1000 ohms per volt, and even this value can introduce errors in some cases.

For example, a 10,000 ohm load resistor shunted by a 50,000 ohm meter (1,000 ohms volt, 50 volt scale) will be reduced to 8,300 ohms approximately, a significant error.

To produce a true 10,000 ohm load in such circumstances it would be necessary to start with a resistor of about 12,500 ohms.

Voltmeter accuracy is the most important factor, since the voltage figure is squared in the formula, and even minor errors will be aggravated. For example, a reading of 9 volts instead of 10 (a 10% error) will result in a figure of 81 being found in the formula in place of the correct figure of 100–an error of 19%.

Valve or solid-state voltmeters can be used, and these would automatically solve any problems concerning sensitivity. However, where a unit is mains operated, with the possibility that one side of the input may be "earthy," due care must be taken to see that this does not conflict with any other "earths" inherent in the amplifier or associated test equipment.

In our own laboratory, we have access to a digital voltmeter and there is no doubt that this makes the job a good deal simpler. Not only is the direct readout free from possible ambiguity, but the inherent high accuracy of these instruments eliminates all doubts as the validity of the final calculations.

Next we require a test signal. The requirements are not unduly stringent. A frequency of 1000 Hz is usually chosen, and the waveform should be a reasonably good sine wave. If a regular audio oscillator is not available the audio output from a conventional signal generator may be used. This usually is about 400 Hz and anything between this figure and, say, 1,500 Hz should be quite satisfactory.

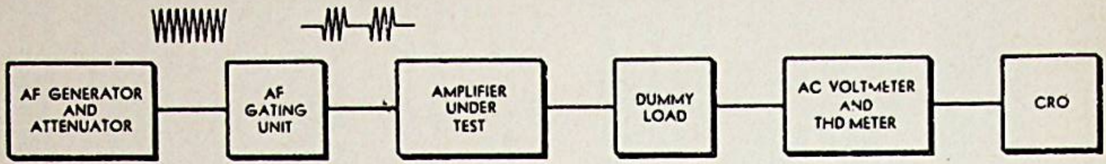


Figure 3

Figure 3. The tone burst method of measuring "Music Power". While better than the system shown in figure 2, it still has some limitations.

Most audio oscillators are capable of generating a reasonably good waveform but anything exhibiting either an obvious peakiness or squared-wave characteristics should be avoided.

A useful refinement to any such audio generator is the provision of a calibrated output. This will enable a sensitivity measurement to be made at the same time as the power input is determined. Together, these two figures give a very clear picture of an amplifier's behaviour.

Finally the CRO. Not a great deal need be said about this since almost any instrument worthy of the name should be suitable. Its job is to indicate the overload point of the amplifier, as shown by just perceptible flattening or other irregularity of one or both halves of the sine wave. Connected across the load, or between the "hot" side and chassis, it will reveal the first signs of overload or clipping, and the need to reduce the signal level or gain to preserve a non-distorted waveform.

When all these items have been connected to the amplifier in the appropriate manner the amplifier is driven up to the overload point, backed off slightly, and the AC voltage developed across the load resistor measured as accurately as possible. From voltage and resistance the power may be calculated according to the following formula

$$W=E^2/R$$

Where:

W is the power in watts.
E is the RMS audio output volts, and
R is the load resistance in ohms.

The foregoing is directed mainly at those who build or design amplifiers and who wish to confirm that an individual unit comes up to the predicted performance figure.

However, there is another situation involving power output measurement; the need to test a commercial amplifier to confirm that it comes up to the manufacturer's specifications. This is a situation which confronts us regularly in our own laboratory. And, while the approach is broadly similar, there are a number of additional factors which have to be taken into account.

These involve such things as the total harmonic distortion for which power output is quoted, the method of rating used by the manufacturer, and so on. Fortunately, quite precise industrial standards have been laid down which provide guidelines for amplifier power measurement.

There are two which are generally recognised. The first and most stringent is that by the American Institute of High Fidelity Manufacturers; the IHF Standards of Measurements for Amplifiers, IHF-A-201, which was published in late 1965, The other standard is the British Standard 3860: 1965.

Other organisations besides the I.H.F.M. have sought to set up standards for the measurement of amplifiers but these have not been as stringent as the above two specifications and the organisations themselves have not gained the same recognition as the I H.F.M.

The method of testing which we have detailed in the first part of the article is similar to, but less stringent than, that described in the American standards under the definition "Continuous Power Output" and in the British standards as "Maximum Power Output.' The main difference is that these standards require that the manufacturer specify a value of total harmonic distortion at which the power is measured, and that the test set-up includes a distortion meter to measure this. (Typical distortion limits are 1% for average high fidelity applications and 0.1% for very high quality amplifiers.) Other terms used in place of "Continuous Power Output" are "Sine Wave Power," "RMS Power" and "Continuous Tone Power.'

More precisely, the British Standard defines ' Maximum Power Output" as follows:

"The maximum measured output power that the amplifier can deliver continuously to a stated load resistance at a frequency of 1000 c/s without exceeding a specified value of (total) harmonic distortion." As explained elsewhere in the standard specifications, ' continuous" implies a test of not less than 30 seconds' duration.

A similar definition and one which has caused some confusion by reason of its similarity is quoted for the term ' Rated Power Output." The difference between these two is mainly one involving the normal spread of tolerances in commercial components, and which results in no two amplifiers, nominally the same, being absolutely identical.

For this reason amplifier manufacturers normally quote a ' Rated Power Output" for their products, and this is meant to imply that the figure quoted is the guaranteed minimum power output which any individual amplifier will deliver. If the manufacturer values his reputation, most units would deliver something more than this value (for the specified harmonic distortion). An individual unit could deliver the same value, but none should deliver less.

In simple terms, the difference between the two terms is the safety margin the manufacturer allows himself in order to compensate for component tolerances.

A block diagram of the equipment used for verifying rated power and determining continuous power output is shown in figure 1.

Another definition often encountered is "Music Power Rating." This seeks to allow for the fact that most amplifiers can deliver a short burst of power which is greater than the continuous power it can deliver. This characteristic is dependent, in turn, on the regulation of the power supply and is considerably lower than the maximum or peak level.

Originally, the I.H.F.M. sought to measure music power with an external regulated supply connected to the amplifier. (Figure 2.) This maintained the supply voltages at the quiescent (no-signal) level regardless of the current drain. The power measurement was made at the reference distortion in the same way as continuous power measurement. While this method did give an approximate idea of the amplifier's performance on music signals it is unrealistic and impractical for several reasons. For one, the dissipation rating of the higher power levels permitted by the external supply.

Further, there is no ripple super-imposed on the regulated supply to add distortion to the waveform near the clipping level, or to add hum to the input signal. Also, apart from such criticisms there is the inconvenience involved in connecting a regulated supply to an amplifier for testing purposes.

For these reasons a new test was introduced by the I.H.F.M. to give a more realistic approximation of the short term power capability of an amplifier. The test involves applying gated sine waves in bursts of 10mS duration at a low repetition rate. (Figure 3.)

The so-called "Tone Burst" test has also been called the "Transient Distortion" test by the I.H.F.M. and the resulting parameter is called the "Dynamic Power" rating. While this test certainly does give a better indication of the short-term power capability of an amplifier it should be realised that an amplifier cannot always deliver this power on the peaks of typical music signals. The actual "music power" of an amplifier at any instant will vary and depends on the immediately preceding amplifier conditions.

In normal conditions the amplifier is not operating at zero output conditions as it does in the tone burst test. For every musical transient that follows a quiet passage there are many which follow moderately loud passages and on these latter transients the amplifier could not deliver the same level of power that it might on the transient following the quiet passage. Thus, an amplifier's music power will lie between the "Dynamic Power" and the "Continuous Power" ratings.

The "Dynamic Power" test has another flaw in that it is intended to indicate the power developed at the same total harmonic distortion as that specified for the rated power output. Distortion measurements on pulsed sine waves are difficult to perform, to say the least. so this forms another region of uncertainty.

In most cases 'Dynamic Power' will be higher by about 20% than the "Continuous Power Output.' If it is any higher it indicates a poorly regulated supply. Amplifiers with an electronically regulated supply will give the same power output on "Dynamic Power' and 'Continuous Power" tests.

The 'Peak Power" rating is one much-favoured by advertising departments in the past but now, fortunately, it is falling into disuse. The people who used it sought to justify it on the basis that the power on a sine waveform is not constant.

The "peak power" figure is arrived at by multiplying the peak voltage (1.4 times the RMS voltage) by the peak current (1.4 times again) to produce a figure twice that which would be produced by using RMS values. The process is about as valid as rating a 100W lamp at 200W peak power. Use of the term "Peak power" has led to the term "RMS power" being coined to distinguish the RMS derived figure from the peak derived figure. However, the term "RMS power" is mathematically incorrect.

We consider the "Continuous Power" test to give the best indication of the quality and capability of an amplifier.

Having explained the various power ratings we can describe the points which must be considered when verifying the manufacturers' specifications of an amplifier. Firstly, as mentioned above the oscillator used in the test must generate a pure sinewave. For accurate distortion measurement the total harmonic distortion should be less than one-fifth of the anticipated distortion of the amplifier under test.

The most important point as far as the actual power measurement is concerned is the accuracy of the voltmeter. While routine checks can be carried out with a multimeter of average accuracy, verification of amplifier performance specifications really requires a voltmeter with an accuracy of 1% of F.S.D.

According to the British standard, the load used for the amplifier should not vary from its nominal value by more than 5%, while dissipating any power up to the amplifier's maximum. The I.H.F.M. standard recommends that it should not vary from its nominal value by more than 1% while dissipating the maximum output of the amplifier, and also that its reactance will not be more than 10% at any frequency up to five times the highest test-frequency. This is an expensive requirement.

While the resistors used for the load must be accurate, care must also be taken to ensure that contact resistance is low in the various connections between amplifier and load. This is particularly important when measuring power into 4-ohm or 2ohm loads.

The mains supply should be within 2% of the mean of the supply range specified by the manufacturer. The British specification further states that tests should be carried out to determine the effect of supply variations over certain limits.

The final stipulation made by the British specification is that all amplifiers should be in use for at least 1 hour before measurements are recorded and that solid state amplifiers should be in use for at least two hours before measurements, to ensure that conditions in the amplifier have stabilised.

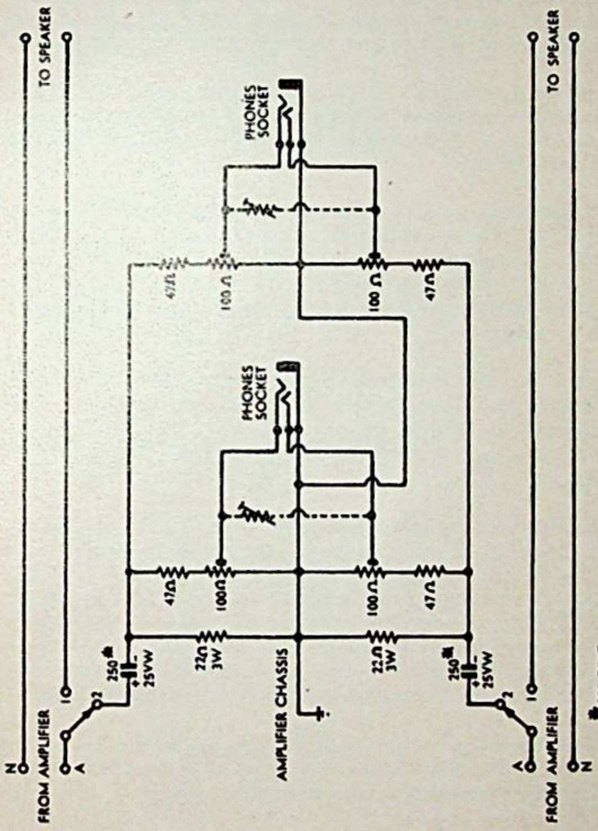
A STEREO HEADPHONE ADAPTOR

This article discusses a universal headphone adaptor which is suitable for all types of amplifier, valve or transistor and all types of stereo headphones, regardless of impedance. Two pairs of headphones can be connected and a switch is provided to silence the loudspeakers.

The only experience that many people have had of headphones is with a pair connected to a shortwave radio or a crystal set. Because the frequency response of such phones is usually peaked in the middle of the range, the sound may sound "clear" enough but it lacks the balance that is essential to high fidelity reproduction. The distortion content is also rather high, as a rule.

In addition, the sound lacks any sense of dimension. In fact, if the individual phones are balanced and connected so that the diaphragms move inwards together and outwards together, the sound appears to originate from a point source right in the middle of the listener's head. It is, in fact, a most peculiar place to have a full orchestra, or a grand organ!

Modern, high fidelity headphones exhibit lower distortion and a response that is much wider and smoother than any of the older, general purpose types. Reproduction, overall, compares very favourably with that from high fidelity loudspeakers.



The circuit of the headphone adaptor.

Furthermore, by feeding the phones separately from the respective channels of a stereo system, the apparent signal source no longer remains captive inside the listener's head.

On fully dispersed stereo program material, the listener has the sensation of sitting right in the middle of the orchestra, with instruments, dispersed on either side, and his head, maybe, inside the lid of the grand piano!

With the more gimmicky "two-channel" type of stereo recording, the listener has the impression of sitting between two distinct groups of musicians. With a "three-channel" type of recording, there is an additional group inside his head!

These impressions are not present when listening to a normal stereo loud-speaker set-up. The sound sources are usually in front of the listener, not adjacent to his ears. Each ear hears each sound source, both by direct and reflection paths, and the listening situation more closely approaches that which it would obtain if the performers were actually located at the far end of the listening room.

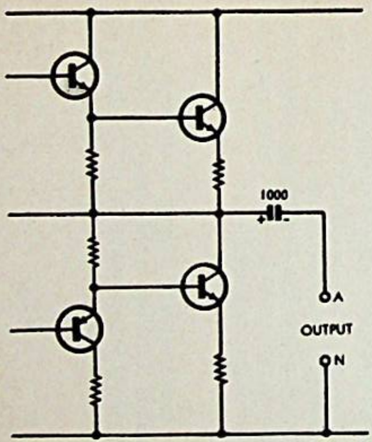
Because of the rather unnatural–though startling–illusions created, it is possible to argue that headphones are not a legitimate method of listening to ordinary stereo program material. Be that as it may, however, the reproduction can be very satisfying and an alternative, far to be preferred to no listening at all.

Headphones do not suffer from apparent attenuation of the high frequencies due to "beaming" effects as do loudspeakers, since the headphone aims right into the ear, as it were. At low frequencies, provided the phones are adequately sealed to the head by flexible surrounds, headphones can produce plenty of undistorted bass; they do not have to set up a large wavefront in a room and cannot excite boomy room resonances. Finally, because the diaphragms need to make only small excursions to move a limited amount of air, they may well contribute less distortion than loudspeakers.

However, the real reason for the continuing popularity of headphones is, as we intimated, that they enable one to listen to music as loud as desired without disturbing the neighbours, babies or parents-in-law.

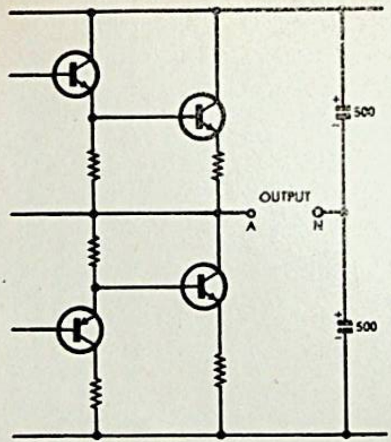
As far as the listener is concerned, headphones cushioned to the ears are far more sensitive than any kind of loudspeakers standing several feet away. This leads to certain immediate and serious complications.

The first arises from the fact that all practical amplifiers have some inherent noise and hum output, even with the volume control turned right down. Through a loudspeaker system this is normally not troublesome but, heard



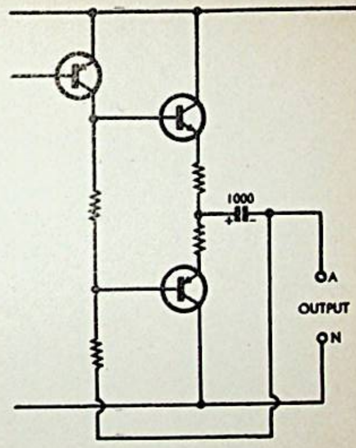
QUASI-COMPLEMENTARY OUTPUT STAGE

Figure 1



QUASI-COMPLEMENTARY WITH CAPACITIVE DIVIDER FOR LOAD COUPLING

Figure 2



COMPLEMENTARY SYMMETRY OUTPUT STAGE

Figure 3

Typical output stage configurations for transistor power amplifiers. A headphone adaptor would need a blocking capacitor in the case of figure 2. None would be necessary for figures 1 and 3, or for valve type amplifiers, which almost invariably use an output transformer.

through earphones, it is generally quite objectionable and sufficient to compromise or ruin enjoyment of the program.

Another aspect is that since so little audio power is necessary to produce adequate output from the headphones, the volume control may have to be set at a critical position, not far advanced from fully off. If, by chance, it happens to be turned up too far, the headphones could easily be damaged.

A possible secondary effect is that, at such low volume control settings, the balance in the two sections may be anything but good, requiring manipulation of the balance control to equalise the two channels.

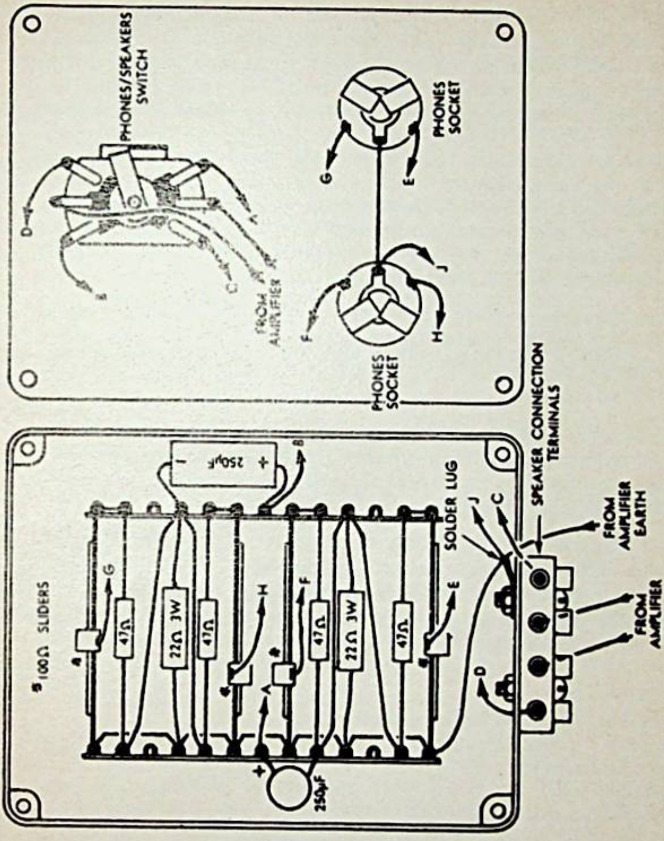
What is clearly required is some kind of attenuation circuit, such that only portion of the voltage at the output of the amplifier ever reaches the headphones. It will reduce the hum and noise fed to the phones and allow the amplifier to be operated with the volume control somewhere near the setting normally employed for loudspeaker listening.

Many commercial amplifiers have jack sockets for one or two sets of headphones and these are fed from aplifier outputs via resistors which may have a value from 150 to 500 ohms. While these resistors may give a suitable order of attenuation for low impedance headphones, e.g., 8 ohms, they will be less effective for the many headphones on the market which have impedances ranging up to 600 ohms, and as high at 10K in one particular case.

What is needed is a voltage divider network which can provide a suitable order of attenuation regardless of the type of headphone used. To this end, we have used a pair of 100-ohm slider resistors fed by 47-ohm resistors for each set of headphones. (See circuit diagram.) By adjusting the moving contact on the slider resistor, the amount of attenuation can be varied over a wide range.

The above network provides only light loading of the amplifier output circuits. While most transistor amplifiers are tolerant of light loading, a few valve types are prone to damage if they are inadvertently overdriven without a load; others may tend to instability. Complementary-symmetry amplifiers (see figure 3) will not operate unless they are loaded, as the load forms part of the bias network for the output transistors.

For the above reasons we have specified a 22-ohm 5-watt resistor as the dummy load for each channel. This value is low enough to ensure correct operation of all amplifiers likely to be encountered and is high enough for the 5-watt rating to be quite adequate for likely levels of operation.



Wiring diagram for the adaptor. Note it may be built to suit only one set of headphones.

In normal stereo systems, the loudspeakers are independent units, each fed by an entirely separate twin lead. The output circuits of the two amplifiers can be entirely independent of each other in respect to both the active and neutral wiring. It is wise to preserve this isolation, at least in respect to the loudspeakers, to minimise any risk of introducing unforeseen complications.

However, isolation cannot be preserved with many types of headphones because they use a common return (or neutral) lead. For this reason we have suggested a 5-wire circuit between the amplifier and headphone adaptor. Four wires provide the separate active and neutral leads for loudspeaker operation; the fifth wire allows the headphones to be connected from the respective activities to chassis.

The use of a common earth return circuit with headphones will cause a complication with those amplifiers which operate with the loudspeaker above chassis potential, i.e., neither side of the loudspeaker is connected to earth, via the chassis.

With amplifiers other than those just mentioned the capacitor may be omitted. With complementary-symmetry amplifiers the capacity must be omitted, since the load forms part of the D.C. bias network for the output transistors, as mentioned earlier.

If there is any doubt as to whether the amplifier on hand should be used with the blocking capacitors, this can be determined without reference to a circuit diagram by measuring the D.C. potential of the loudspeaker voice-coil circuit with respect to the amplifier chassis with no signal applied. If it is zero, then no capacitor is necessary.

Parts List

1 Metal box, 4-5/8 x 3-5/8 x 2-1/8 inches.
1 two-pole, two position switch.
2 Stereo jack sockets.
2 13-lug tagstrips.
1 4-way insulated terminal block.
2 250uF/25VW electrolytic capacitors (see text).
2 22-ohm, 5-watt resistors.
4 47-ohm, ½-watt resistors.
4 100-ohm,slider resistors.
1 knob, 4-conductor cable with shield, cord clamp, solder lug, hook-up wire, screws, nuts, etc.

Note that if the D.C. potential of the voice coil proves to be negative with respect to the chassis, as was the case with one or two circuits we have seen in overseas magazines, the polarity of the electrolytic capacitors in the adaptor unit would have to be reversed to that shown in the circuit and wiring diagrams.

Problems to do with the internal circuity or amplifiers are mainly confined to transistor types. Valve type amplifiers almost invariably used output transformers with no D.C. potentials associated with the secondary winding; in most cases, one side of the secondary winding was earthed to chassis. No blocking capacitor is necessary in a headphone adaptor but, by the same token, the presence of a blocking capacitor would not adversely affect its operation.

The prototype headphone adaptor was constructed in a die-cast metal box, which is available from most electronic part suppliers. All the components are installed between two 13-lug tagstrips. The switch to select loudspeakers or phones is a two-pole, two-position type. The wiring layout is not critical but should be neat and tidy and follow good wiring practice.

The cable to the earphone adaptor needs five effective conductors and could logically be five insulated leads inside an outer covering. Because it was most readily available, we used a cable with four insulated conductors, two of which have a common outer braided shield; this was used as the fifth conductor and logically as the earth lead from the amplifier chassis.

The four insulated leads are used for the active and neutral loudspeaker leads; the neutral lead is that lead which is at zero A.C. potential with respect to the amplifier chassis, i.e., the earthy side. On these amplifiers with screw terminals for the loudspeaker connections, the active side may be coloured red or coded with a "plus" sign. If this is not the case, the active side of the speaker output may be determined by trial and error when the adaptor is completed; if the headphones are effectively connected between neutral side of the amplifier output and the amplifier chassis, no sound will be heard.

The cable enters the case through a grommetted hole and is clamped to avoid risk of straining the connections. The shield is terminated to a solder lug which is secured by the same screw which secures the cord clamp. The neutral lead from the amplifier are then brought out through the grommetted hole and terminated on the four-way insulated terminal block. The two leads from the "speakers" lugs on the speakers/phones switch (terminal 1) are also brought out through the grommetted hole and terminated on the four-way terminal block. The four leads from the loudspeakers can then be taken from the other side of the terminal block.

Alternatively, the terminal block may be dispensed with and the four insulated leads in the cable used to take the active leads from the amplifier to the speakers/phones switch and back to a terminal block on the rear panel of the amplifier.

Once the adaptor has been completed the slider resistors should be set to the position which would give minimum volume in the headphones. After having connected the various leads to the amplifier and connected the loudspeakers to the appropriate points in the adaptor circuit, program material should be played through the loudspeakers with volume, balance and tone controls at their normal settings. Leaving these set, switch to the headphones and adjust the appropriate slider resistors for a suitable level in each channel. With the levels set in this way, switching from speakers to headphones can be done without the need for resetting the volume control. The effective signal/noise ratio should be about the same.

The diagrams provide for operating two pairs of headphones but, if provision has to be made for only one pair, one of the sockets may be omitted, along with the associated 47-ohm and 100-ohm slider resistors.

Readers may care to experiment with the idea of cross-coupling the two headphone circuits to diminish deliberately the isolation between the two channels.

This has been the subject of a fair amount of discussion in overseas audio journals. The circuits usually involve various configurations of L, C and R, intended to make the cross-coupling frequency and phase conscious, in an effort to simulate the conditions which obtain in ordinary loudspeaker listening.

Before being qualified to offer an opinion about the various circuits which have been suggested, it would be necessary to conduct quite lengthy listening tests with a variety of stereo program material and this we have just not had time to do. It is obvious, however, that provision of even the simpler kind of network would add materially to the cost, bulk and complexity of a headphone adaptor.

For those who wish to experiment on a simpler basis, however, low value resistors or a potentiometer may be tried in the position shown dotted on the circuit diagram.

Go out and purchase yourselves a pair of headphones then, build this adaptor unit and you can listen to stereo any time you like.

PHASING STEREO LOUDSPEAKER SYSTEMS

One of the important requirements in a stereo amplifier system is that the loudspeakers should operate in phase with each other. This can be ensured by paying proper attention to the connections and the wiring. It can be double-checked by a simple test.

There is nothing very mysterious about the term "in phase", as applied to loudspeakers in a hi-fi system. It means that, if fed with an identical signal, all cones in the system would move in the same direction – either towards the listener or away from the listener. (See figure 1).

In a six-loudspeaker system, with a high-frequency "tweeter", midrange and low frequency "woofer" unit for each channel, there will be six cones in phase. With a four-loudspeaker system, there will be four cone assemblies in phase and, with a two loudspeaker system, there will be two.

The situation depicted in figure 1 can be defined as the normal one for a stereo system, on the assumption that the original sound image will be recreated most accurately when the loudspeakers are arranged in two compact groups, operating in phase, directed towards the listening area, and with the groups separated by a distance appropriate to the dimensions of the room.

To be strictly practical, the phase relationship of a high frequency tweeter to the mid- and low-range units, or to its counterpart in the other channel, is not as critical as the foregoing paragraph might suggest. This is due partly to the fact that separate tweeters must, of necessity be separated from other loudspeakers by a distance which is significant in terms of the wavelength of the sound being propagated; how the wavefronts interact is as much a matter of cone location as of cone phasing. In addition, what the listener hears is greatly modified by random reflections from the walls, floor and ceiling of the listening room.

Appreciating this fact, some system designers deliberately point tweeter loudspeakers upwards on the basis that high frequency sound dispersed (or bounced) from wall/ceiling corners is more acceptable than that beamed directly at the listeners. Strangely, however, such designers usually pay lip service to phasing by seeing to it that tweeter cones move outwards in their housing at the same instant that other cones in the system are doing likewise!

With loudspeakers handling the mid-range of frequencies – say 400 to 4000 Hz – reflections can still disguise the effect of in-phase or out-of-phase operations to some extent. It is nevertheless reasonable to set down, as a positive requirement, that mid-range loudspeakers should operate in phase with their associated woofer and with their opposite channel counterparts in a stereo system.

There can be no equivocation, however, on the fact that the cones producing the low frequencies in each stereo channel be in phase with each other. While there may be a considerable difference between the middle- and high-frequency content of a pair of stereo signals, the low frequency components in the two channels may be quite similar in amplitude and phase. Particularly is this likely to be so in recordings which have been made using a single stereo microphone or for compatible stereo/mono playing. In the case of mono records played on a stereo system, the content will, of course, be the same.

In all these cases, it is essential that the cones handling the low frequencies co-operate in the effort to move air towards or away from the listener on the respective half-cycles.

If the bass cones move out of phase, they will tend to pump air back and forth in the space between them instead of propagating the sound out into the listening room (Figure 2). The apparent effect, upon the listener is that the system lacks bass response.

Many of the uncertainties about loudspeaker phasing arise from the fact that enthusiasts often do not start to worry about it until the systems are virtually ready to go. The loudspeakers are installed in an enclosure, wired up and linked to a couple of unmarked connectors on the back. Non-coded wires are used for the run back to the amplifier and only then – if he is aware of the requirement at all – does the enthusiast pause to wonder about whether or not the loudspeaker will operate in phase.

Uncertainties about phasing can largely be obviated by taking appropriate measures at the time a system is being set up.

The first requirement is an ordinary 1.5V torch cell, a pair of clip leads and a pair of eyes. Try touching the cell across the voice coil terminals and note the connection which causes the cone to move forward in its housing when the contact is made. Having discovered the appropriate connection, mark the loudspeaker with the same polarity as the battery – plus and minus (Figure 3).

In many modern loudspeakers, the polarity will already be marked but with the risk, with some imported loudspeakers, that the opposite convention will have been followed. By checking and marking each individual loudspeaker yourself, any worries on this score can be avoided.

Having marked the polarity of all loudspeakers, combinations involved for each channel (woofer, mid-range, tweeter) can be interconnected using two differently coloured wires. The idea is to assume that one side of the

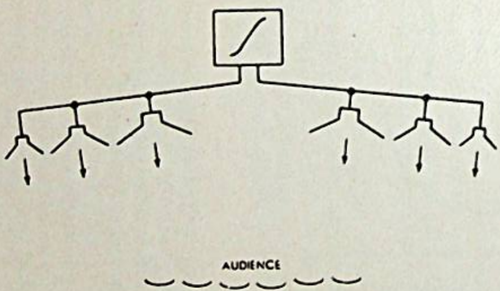


Figure 1: Loudspeakers are said to be connected in phase if their cones move in the same direction when the voice coils are fed with a common signal.

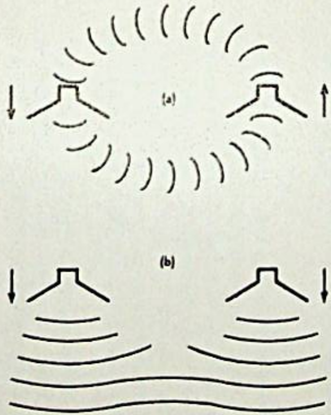


Figure 2: When the low frequency cones in a stereo system operate out of phase (a) they circulate the air in the space between them. In phase (b) the energy is propagated into the room.

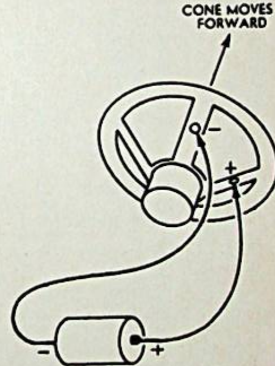


Figure 3: Observe which way round a 1.5V cell has to be connected to a loudspeaker to cause its cone to move forward in the housing. Mark the voice coil connections to correspond.

input will be "earthy" and will normally go to the "negative" of each loudspeaker in the group. The other or "active" wire, will go to the positive lug(s), either directly, or through series inductor(s) and/or series capacitor(s) where these are called for.

Having this interconnected the loudspeakers, polarised twin lead can be brought out and run back to the amplifier,with one specific colour being selected to represent the "positive" or "active" input to each loudspeaker system.

The next step is to examine the loudspeaker connections to the stereo amplifier. Usually, these will be distinguished in some way. They may be lugs with earth or positive signs, coloured terminals or a polarised plug and socket.

If they are completely unmarked, it may be necessary to examine the circuit and the wiring to establish how the terminals relate to it. In a valve type amplifier, one side of the output transformer secondary will normally be connected to the chassis and this can be marked arbitrarily "earth"; the other will be "positive". In a transistor amplifier the output circuits may connect more directly to the transistors and it will be necessary to identify the equivalent connections in each channel.

In the latter case – or in either case – the nomination of particular terminal connections as active or positive may be quite arbitrary, provided the SAME convention is applied to BOTH amplifier channels.

Either way, whether the connections are clearly marked or have to be traced out, the requirement is that the two-colour-coded loudspeaker leads be identically connected to the two output channels, the leads going to the equivalent places in the respective output circuits.

In normal stereo amplifier systems, the two channels are identical and, if pre-phased loudspeakers are connected to them symmetrically by colour-coded leads, it is a virtual certainty that the loudspeakers will operate in phase when the system is fed from a mono source, such as a radio turner, mono tape recorder, or pickup cartridge paralleled for mono operation.

The remaining requirement is to ensure that properly phased signals reach the amplifier from a stereo cartridge. Most modern stereo cartridges have four output lugs which can be identified by markings or by manufacturer's data as left channel active and earth, and right channel active and earth. Sometimes, the two "earth" pins are already interconnected and/or bridged to the cartridge case. It is simply a matter of checking through the pickup wiring and connectors to ensure that the two 'actives" go to the two active input connections of the two

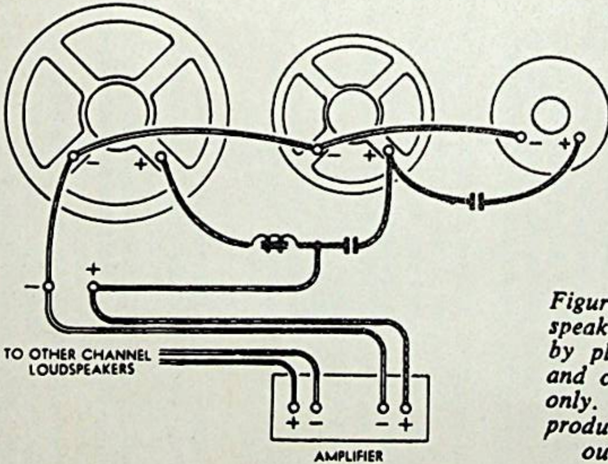


Figure 4: Having marked the polarity of all loudspeakers as per figure 3, they can be connected in phase in each system and wired back to the amplifier by colour-coded leads.

Figure 5: The phasing of loudspeaker systems can be checked by placing them face to face and changing one set of leads only. The in-phase connection produces louder sounds than the out-of-phase connection.

amplifier chains and that the "earths" connect to the appropriate earth returns.

Provided that these precautions are taken it is almost inevitable that the sound emanating from the respective loudspeaker systems will have the correct phase relationship.

But you still aren't quite convinced? You would like to verify the inevitable. Well there's an easy way of doing so, at least for those whose loudspeakers are mobile.

Take the loudspeaker systems and place them almost face to face, forming a narrow V with the top of the V towards the listening position. Leave the system set up to play stereo, but put on a mono disc. Select a track that you know well and set the volume so that it is about the level you like to listen to. The treble may be down a little because of the strange orientation of the loudspeakers, but it should sound acceptable enough.

Listen carefully so that you will remember the tone and volume, leaving the controls etc. Switch off and reverse the leads to one loudspeaker system only, switch on again and play the same track. There will be a noticable difference in volume and bass response. Assuming that the original phasing was correct the sound should now be much lower and noticeably lacking in bass because, instead of combining to propagate the sound out the open V, the cones will be merely pushing the sound back and forth across the intervening space. If the original phasing was wrong, the new condition should be a marked improvement in volume and bass response.

Select the condition which gives the fullest sound – the original connection is the phasing instructions have worked out!

A similar test with a radio or other mono input should produce the same result and confirm that the system is correctly phased for all signal inputs.

Incidentally, the "Stereo-Reverse" position on the function switch of many amplifiers does not effect phase; it simply switches the signal belonging to the right channel to the left, and vice versa. Just a few amplifiers have been fitted with a phase reversing switch, usually on the back, to change the phase to one loudspeaker. This should be very carefully examined and marked so that its "normal" position is known. If it is not so marked it can be a constant source of confusion – and a constant temptation to "experiment".

Where loudspeaker systems are not mobile, the problem of verifying phase is more difficult, being rendered the more so in a room which is naturally "live" or in which multiple echoes are created by clusters of hard furniture, passages, etc.

Perhaps the best test is a voice from a mono source, or simple easily recognisable sound effects. Listen critically to the sound, then swap one pair of leads over only and listen again. Theoretically, at least, the in-phase connection should provide the more definite sound image, midway between the loudspeakers.

Some special test and demonstration records contain tracks which are intended to assist the listener in verifying phase, how to use the tracks being the subject of an appropriate instruction on the jacket. However, all these tests rely heavily on listener perception and can be clouded by room acoustics.

Which brings us back to the original point of this article. The best way to cope with phasing is to start out with a battery, a marking pencil, polarised connectors and colour coded wiring and to beat the problem by eliminating it!

HIGH IMPEDANCE 4-CHANNEL MIXER

Here is a Four Channel Mixer, embodying field effect tansistors. Boasting input impedances of 5M it is capable of working with high impedance crystal and ceramic sources while maintaining the bass response inherent in these devices. It will mix up to four inputs with a total control interaction of only 1dB.

The microphone preamplifiers each employ a field effect transistor in a conventional configuration at the input, with magnification of the gate resistor by overall negative feedback. Each pre-amplifier has an input impedance of about 5M ohms with a gain of 2.8. However, the additional gain of the mixer stage increases the overall microphone gain to 130, giving a nominal sensitivity of about 2mV.

If it is intended to use only dynamic microphones the preamp inputs may be shunted by 47K at the mic. socket. This will improve the signal-to-noise ratio. However, at –48dB we think that the noise level is quite acceptable.

Those readers who consider that a signal-to-noise ratio of 48dB is unacceptable may use lower impedance preamplifiers providing that they do not require facilities for crystal microphones. In this case the signal-to-noise ratio will be 54dB. However some changes to component values are required so that the preamplifiers can be operated from 9V.

The base biasing resistor for the first transistor of each microphone channel (T1 and T2) should be changed from 180K to 390K, also the collector resistors of T5 and T6 from 22K to 10K. The mixer stage should be of the current design, except that the series input resistors to the mixer stage from the microphone preamplifiers should become 220K.

The mixing stage uses a single NPN silicon transistor type 2N3565 or BC108 or similar. The stage is a conventional common emitter configuration with stable gain, low distortion and low input/output impedances provided by negative voltage feedback. The basic circuit is shown in figure 1.

The low impedance level produced at the input (point "A" in figure 1) due to the feedback through resistor Rf plays an important part in the operation of the mixer stage. The low impedance ensures that the gain of the stage as seen by any of the inputs is virtually independent of the other terminals. This means that the signals can be derived from standard gain control potentiometers without appreciable interaction between the controls.

The input impedance of each input is very close to the value of its series input resistor-R1, R2, or R3, etc. Similarly the gain of the mixer stage, for each signal, is given by the ratio between Rf and the series input resistor; signal 1 receives a gain of Rf/R1 signal 2 a gain of Rf/R2, and so on.

In practice however, the value of the Rf is modified by the shunting effect of the collector-to-base bias resistor. Where the latter is high relative to Rf, as in the 1966 design, it can be ignored.

In the present design the value of the collector-to-base bias resistor is 1M and the shunting effect cannot be ignored. As Rf is also 1M the total value of feedback resistance is the parallel-sum of 1M and 1M, that is 500K ohms.

The maximum number of signals which may be mixed together with this basic circuit depends upon the performance required, but is fundamentally determined by the gain of the stage before feedback is applied. For a given open loop gain the maximum number of inputs that can be used depends upon the input impedances and gains required for the signals, together with the noise, distortion and control interaction levels which can be tolerated.

In general, the higher the gains required for a given value of feedback resistor, the fewer the inputs which may be connected for a given set of noise, distortion, and control interaction criteria. Thus with the basic circuit shown in figure 1, excellent performance can be obtained when mixing up to 10 medium impedance inputs. However, due to the above design considerations – which indirectly also control the overall gain of the system – this number was not attempted in this design.

In fact the number of channels is only four, two for microphones and two for pickups, which gives a control interaction of 1dB. Even so, any combination of microphone and pickup inputs may be used. So long as the parallel sum of all the series resistors, to the mixer stage, is not less than 22K, the control interaction level will be slightly less than 1dB.

However, if slightly more control interaction can be tolerated two additional channels may be added, either two microphone, two pick-up, or one of each. In this case the total control interaction will be 1.5dB, an increase of only 0.5dB.

If more microphone channels are required yet with a low control interaction level, the earlier dynamic microphone preamplifier, suitably modified, could be added. This will provide a total of two crystal microphone and two dynamic microphone channels together with two pickup channels, the total control interaction being about 1dB.

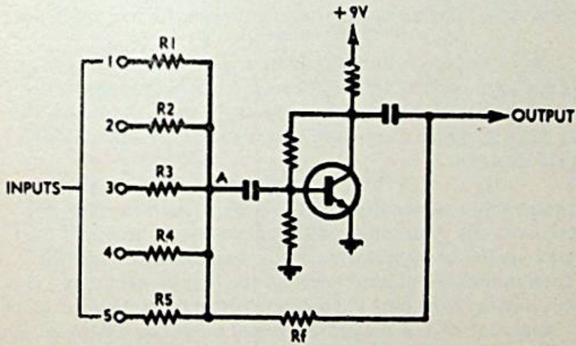


Fig. 1

The basic mixer circuit. The factors governing the circuit's behaviour are discussed in detail in the text.

In the mixer stage of the present design the absolute value of feedback resistor is 500K and the input resistors for all stages are 82K, giving a mixer stage gain of about 4.6 times. Thus the nominal sensitivities, for a 250mV mixer output, are: microphone input 2mV, pickup input 60mV.

The metalwork and hardware is basically the same as that of the original mixer unit. As may be seen from the inside photographs the unit is compactly built into the 6½in x 4½in x 2in metal box which has been used previously for a number of projects. The box has a "biscuit-tin lid" front panel.

The four gain control potentiometers are mounted in a row along the front panel, and the only other item on the panel is an "on-off" slide switch. The long side of the case furthest from the controls is used to mount the four input connectors and for the output cable entry. Four rubber feet complete the unit.

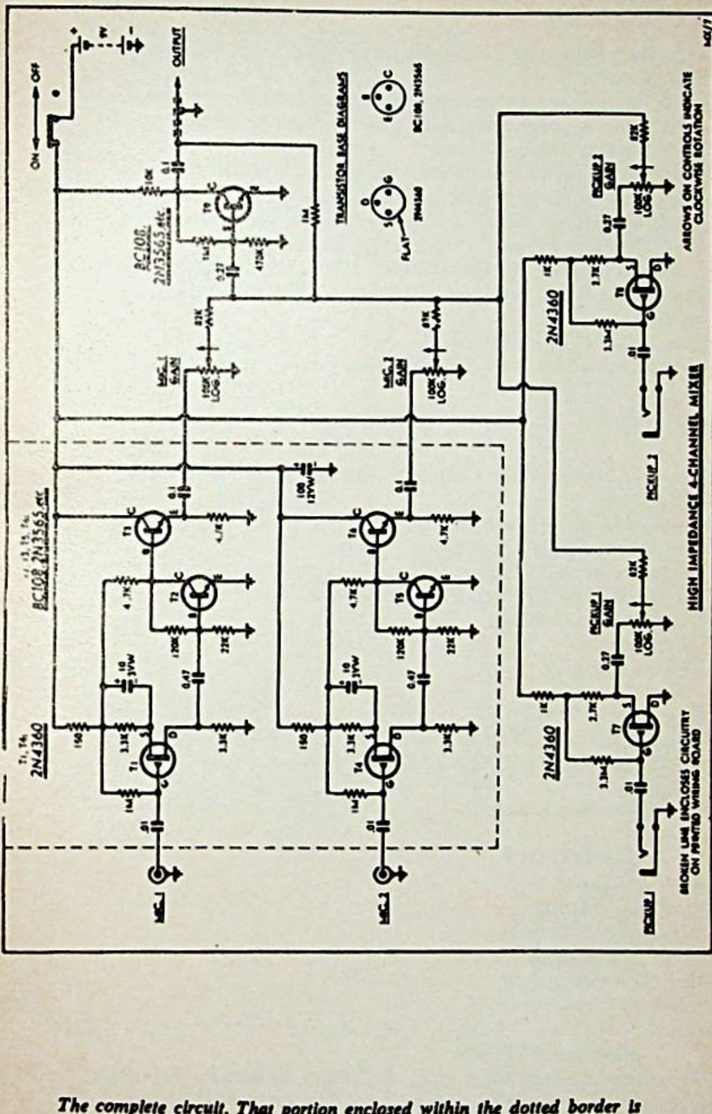
The pickup and mixing circuitry is wired on two strips of miniature resistor panels each 10 lugs long. Each section may be wired and individually mounted, and the interconnections completed after mounting. The small 9 volt battery is mounted under the cable entry by a small bracket bent from scrap aluminium; a small "C" clamp made of the same material is used to retain the signal output cable.

Construction of the mixer unit should be a relatively straightforward task even for beginners, as we have provided a complete wiring diagram. Using this in conjunction with the photographs there should be no difficulty in producing a close copy of the prototype and obtaining similar performance.

The suggested order of assembly is as follows: After preparing the metalwork, wire the preamplifier components on the printed board and the pick-up/mixer components on the two miniature resistor panels. Then mount the printed board in the case by means of 1-8 in Whitworth screws, using nuts to space the board from the bottom of the case. Note that two of the printed board mounting screws are used to fasten the rubber feet to the case.

The two miniature resistor panels may now be mounted in the case using 1-8 Whitworth screws with additional nuts to space the panels about ½in from the case bottom. Having fitted the wiring boards, the signal connectors may now be inserted, together with the battery and output cable.

With the mechanical assembly complete it is a simple matter to complete the interconnections between the wiring boards, input connectors, and the front panel controls. The interconnections are straightforward and if the wiring diagram is followed carefully they should present no difficulty.



The complete circuit. That portion enclosed within the dotted border is built on a printed wiring board.

SPECIFICATIONS

Four-channel audio mixer unit using silicon field effect and bi-polar transistors.

Two high input impedance microphone channels for crystal and dynamic types, input impedance 5M with a gain of 130. Response 35Hz to 25KHz within 3dB.

Two crystal/ceramic P.U. channels; Input impedance 5M, with a gain of 4. Response 35Hz to 25KHz within 3dB.

Designed for nominal 250mV output level but will deliver more than 3V RMS before clipping. Output impedance less than 10K.

Mic. channels noise 48dB below 250mV output; pickup channels better than 60dB. Total harmonic distortion at 1V RMS output 0.1 per cent.

Maximum control interaction 1dB.

Operates from one small 9V battery with a current drain of 9mA.

PARTS LIST

1 Metal case, 6½in x 4½in x 2in, with "biscuit-tin-lid" front panel.
1 Wiring board.
2 Miniature Resistor Panels (10 lug).
1 Slide switch, 1 pole 2 position.
4 Transistors, type 2N4360
5 Transistors, type 2N3565, BC108 or similar.
2 Microphone connectors.
2 Pickup jacks.
1 Shielded jack plug.

RESISTORS

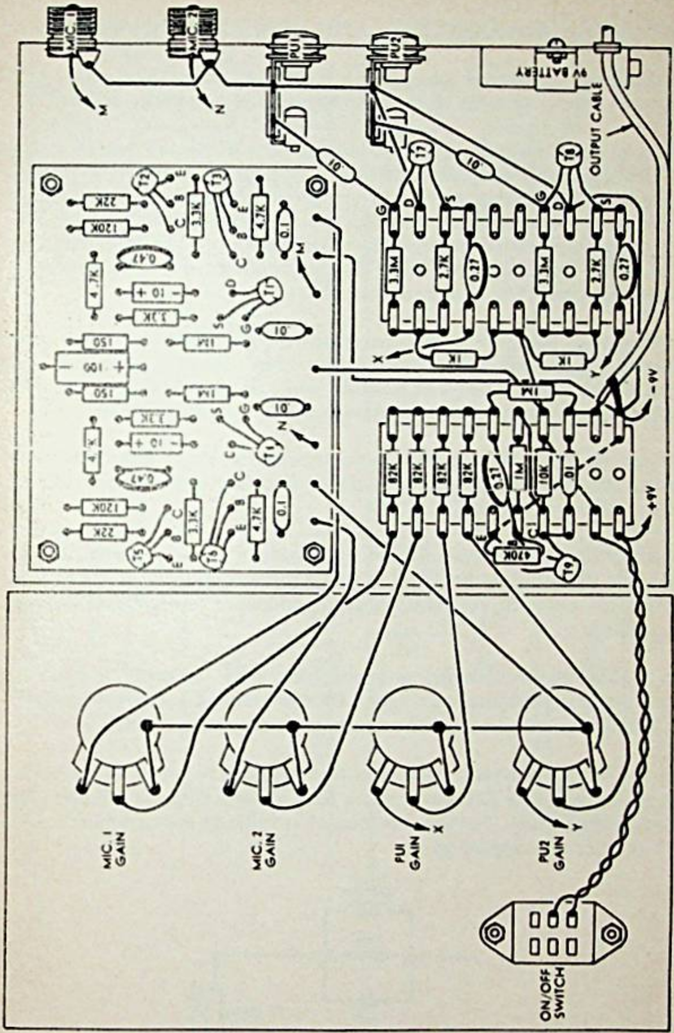
Half-watt, 5 per cent: 2 x 150 ohms, 2 x 1K, 2 x 2.7K, 4 x 3.3K, 4 x 4.7K, 1 x 10K, 2 x 22K, 4 x 82K, 2 x 120K, 1 x 470K, 4 x 1M, 2 x 3.3M.
4 100K log potientiometers.

CAPACITORS

2 .01uF LV plastic.
3 0.1uF disc ceramic.
3 0.27uF disc ceramic.
2 0.47uF disc ceramic.
2 10uF 3VW electrolytic.
1 100uF 12VW electrolytic.

MISCELLANEOUS

1 9V battery and connector; scrap aluminium for battery and output cable clamps; 4 x rubber feet; length of shielded cable for output lead; connecting wire; 4 knobs; nuts, bolts, solder, etc.



This wiring diagram will assist with the general layout, resistor panel wiring, and general interconnections.

SPEAKER GAIN CONTROL AND CONTOUR NETWORKS

For constant impedance gain controls for use in voice coil circuits, and for a speaker contour network to smooth the response of a peaky speaker.

The gain control was developed to control the volume of a pair of remotely located stereo speakers operating from a stereogram. Constant impedance control was decided upon as I was not certain of the reaction of the input transistors to variable loads. The fact that I had a two-pole, five position switch on hand settled matters. The volume is dropped in 3dB steps. Standard ½W resistors were used. The suggested versions for 8, 4, and 2 ohms might be better with heavier resistors.

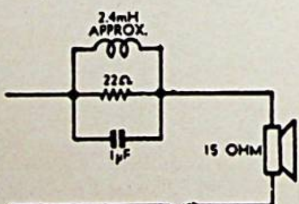
In the case of the lower impedance systems–particularly the 2 ohm system–resistance losses must be considered. The cables should be kept as short as possible and made as heavy as possible, while the switch should be a good quality unit in which contact resistance will be consistently low.

The second idea is for a speaker contour network. This started when I was given a Mini Speaker enclosure. I decided to check how things would sound with a twin cone speaker.

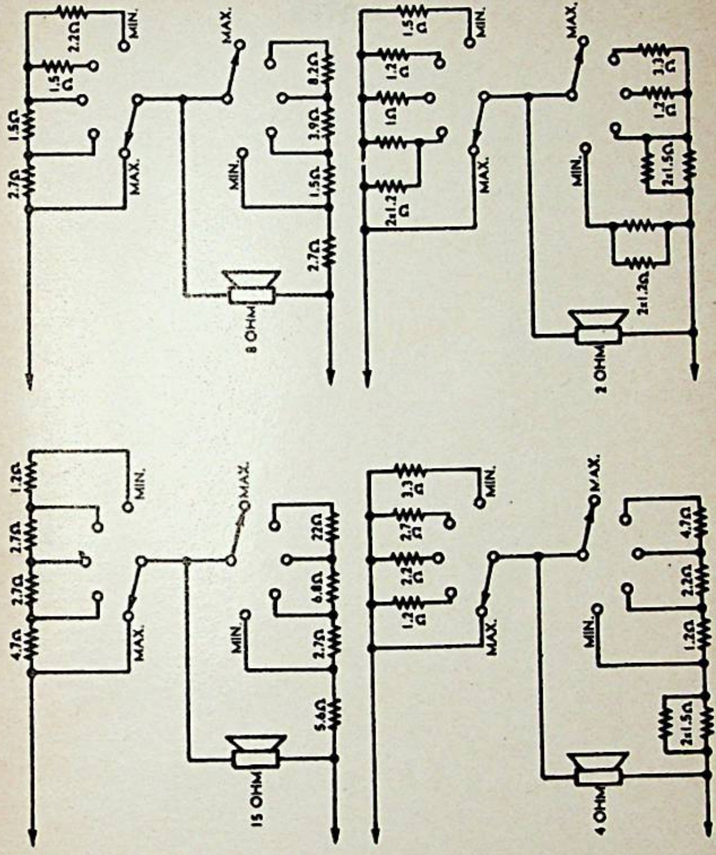
Results were not bad, but sounded a bit strident due, apparently, to the usual rise in output in the 2KHz to 4KHz region. I decided to try an idea, a damped LC network, resonant about the middle of this range, in series with the speaker.

My coil former was 1¼in diameter and 7/8in wide. (Actually, it was a plastic spool that had held insulation tape.) On this I wound about 160 turns of 16 S.W.G. enamelled wire. Inductance measured about 2.4mH.

The speaker sounds very much smoother with this network. In fact, there is not much to choose between it and a Rola speaker fitted in a Briggs type drainpipe enclosure. The latter is somewhat fuller at the bass end, but there is little difference higher up.



A contour network to control a peak in the 3KHz region.



Constant impedance gain controls to suit four voice coil impedances.

NOTES

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