

PRACTICAL STEREO HANDBOOK

THE COMPLETE GUIDE TO STEREO

By Clive Sinclair



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**PRACTICAL
STEREO
HANDBOOK**

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BOOK I

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We invite all authors, whether new or well established, to submit manuscripts for publication. The manuscripts may deal with any facet of electronics but should always be practical. Any circuit diagrams that may be included should have been thoroughly checked by the author. If you are considering trying your hand at writing this type of book we suggest that you let us have a short summary of the subject you intend to cover. We will then be able to let you know the size of book required and perhaps give you some advice on presentation.

PREFACE

This book has been written in the hope that it will serve as a useful reference for all those who either possess or intend to possess stereophonic equipment either in the form of a tape recorder or stereophonic records. A considerable amount of confusion has arisen about the various methods of recording and reproducing sound stereophonically so we hope we can provide a better understanding of the various systems now in use.

Although people have experimented with stereophonic sound for a good many years, the commercial possibilities have only recently been exploited and in the next few years new techniques will probably result in the virtual extinction of monophonic sound equipment and stereophonic radio broadcasting will become the rule rather than the exception.

We think you will find this and subsequent Bernard's books on stereo, right up to date and if you have any queries please write to us. Even if you do not wish to ask us anything we would be very pleased to hear your comments on these books for they are written for you not for us, and only in this way can we know what you want.

CLIVE SINCLAIR

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CHAPTER I

STEREO LANGUAGE

Before it is possible to discuss stereophonic sound with any simplicity it is necessary to understand all the various terms that are commonly used. These are listed below with a complete explanation of each term.

Audio Reproduction

1. *Monaural and monophonic*: these two terms are synonymous. The term monophonic having been coined as a counterpart to stereophonic. They both mean audio information on a single channel. For example, the blocking of one ear means that a monophonic signal is received by the brain.

2. *Binaural*. This term refers to reproduction of two sound channels by earphones, one for each channel. The two channels are recorded by two microphones separated by means of an intervening partition representing the head of a listener. The microphone is designed to have a similar response to that of the ear.

3. *Stereophonic*. Audio information carried on two or more sound channels intended for reproduction by a similar number of speaker systems. Unlike stereo photography where only two channels are used, the more channels used in stereophonic reproduction the better the result so long as the various techniques required are taken into account.

4. *Pseudo Stereo*. Systems have been developed which produce from a single channel source some of the qualities associated with stereophonic sound. The simplest method is to feed a single channel source to two speakers placed several feet or more apart. This however, does not usually give a very satisfactory

result. Another method is to acoustically delay all the frequencies by a fraction of a second by passing them through a long tube before feeding them to the second speaker. In the case of the reproduction of an orchestra, the violins being on the left and the bass instruments on the right, it is often possible to produce a pseudo stereo result by feeding the bass to a speaker on the right and the high frequencies to a speaker on the left. Alternatively, some form of three-channel effect may be obtained by feeding the bass to a centre speaker and the high and middle frequencies to the left and right respectively. Other devices operate electronically, achieving a time delay which varies with frequency. This tends to have the effect of spacially distributing the various orchestral instruments.

5. *Coded Stereo*. This system, which will probably be used more and more as the art of stereophonic sound is improved, consists of a single channel audio accompanied by a subsonic code signal which controls the volume of sound fed to the speakers on the left, right and centre. The subsonic signal causes an appropriate amount of signal to be fed to the correct speaker at the right time. Any number of channels may thereby be formed. This system has been used for some time in certain cinemas.

Stereo on Gramophone Records

1. *Dual groove record*. The dual groove record uses two separate sets of grooves, one for each channel. It requires two completely separate cartridges side by side. The main problem is the alignment of the cartridges

which must be very precise otherwise the two channels will be reproduced one after the other and the effect will be lost. This method is not used commercially nowadays.

2. *Single groove recording.* This employs a single set of grooves for both channels and requires only one cartridge for stereo playback. There are two basic methods. The first employs both vertical and lateral modulation of the groove. The second, employing only lateral modulation in the manner of a monaural record together with a modulated carrier frequency. It may however be played back on a 45/45 cartridge which has two elements—one responding to stylus motion at an angle of 45° to one side of the vertical and the other element responding to stylus motion at a 45° angle to the other side of the vertical. The sum frequency of the two sound channels $a + b$ is cut laterally, the difference frequency $a - b$ is cut vertically. This is done at a much reduced level. In playback the 45/45 cartridge acts as a matrixing device so that one of the elements delivers essentially an 'A' signal and the other delivers essentially a 'B' signal.

3. *A 45/45 record.* In this form the record groove is in the form of a V each wall of the V being at 45° to the vertical, so that the angle between the two sides is 90° . The left wall is recorded so that it contains channel 'A' information for the left speaker, the other wall contains channel 'B' information for the right speaker. The signal for the left speaker causes the stylus to move at a 45° angle to vertical namely from bottom left to top right, i.e., left side is cut in a manner that causes the stylus to move slantwise along the right wall. The right channel causes the stylus to move from bottom right to top left along the left wall. The combination of signals from both channels causes the stylus to move in some intermediate position. The cartridge employed for playback, the so called 45/45 cartridge, contains two elements, one responding to a stylus motion of 45° to right of vertical and the other corresponding to stylus motion at an angle of 45° left of vertical.

Stereophonic Recording on Tape

1. *The heads.* In the case of stereophonic tape recording two separate heads are used, these may be either in line or staggered. In the case of the in line head two tape heads in a single casing are mounted directly above one another so that their gaps are in exact vertical alignment. If the stereo tape runs from left to right the upper head reproduces or records the left channel and the lower head the right channel. The in line playback head is suitable only for recorded stereo tapes with one channel directly above the other—it is not suitable for staggered tape. With a staggered head, separate heads are spaced about $1\frac{1}{4}$ inches apart for playing or recording the upper and lower halves of a stereo tape. If a tape runs from left to right the head on the right is for the right channel and operates on the lower track of the tape. Staggered heads are suitable only for recorded tapes with tracks staggered in a corresponding fashion. Staggered heads and staggered tapes are now no longer used.

2. *Balance Control.* This is a device which is incorporated on all but the simplest stereo amplifiers to vary the volume of each speaker system relative to the other, at the same time however, maintaining their combined volume virtually the same. As one speaker increases, the other decreases in volume, the sound appears to shift from left to centre to right or vice versa. The adjustment is varied so that the correct effect is produced. Another name for Balance Control is focus control.

3. *Master Gain Control.* This is a device on most stereophonic amplifiers which simultaneously controls gain of both channels. The master gain control should not cause a difference between channels of more than 1 or 2db at any point.

4. *Phase Reversal Switch.* A device on a stereo amplifier or even in the speaker system for shifting the phase by 180° on one channel. This usually means merely interchanging the two leads to one of the speaker systems. If stereo speakers are improperly phased relative to one

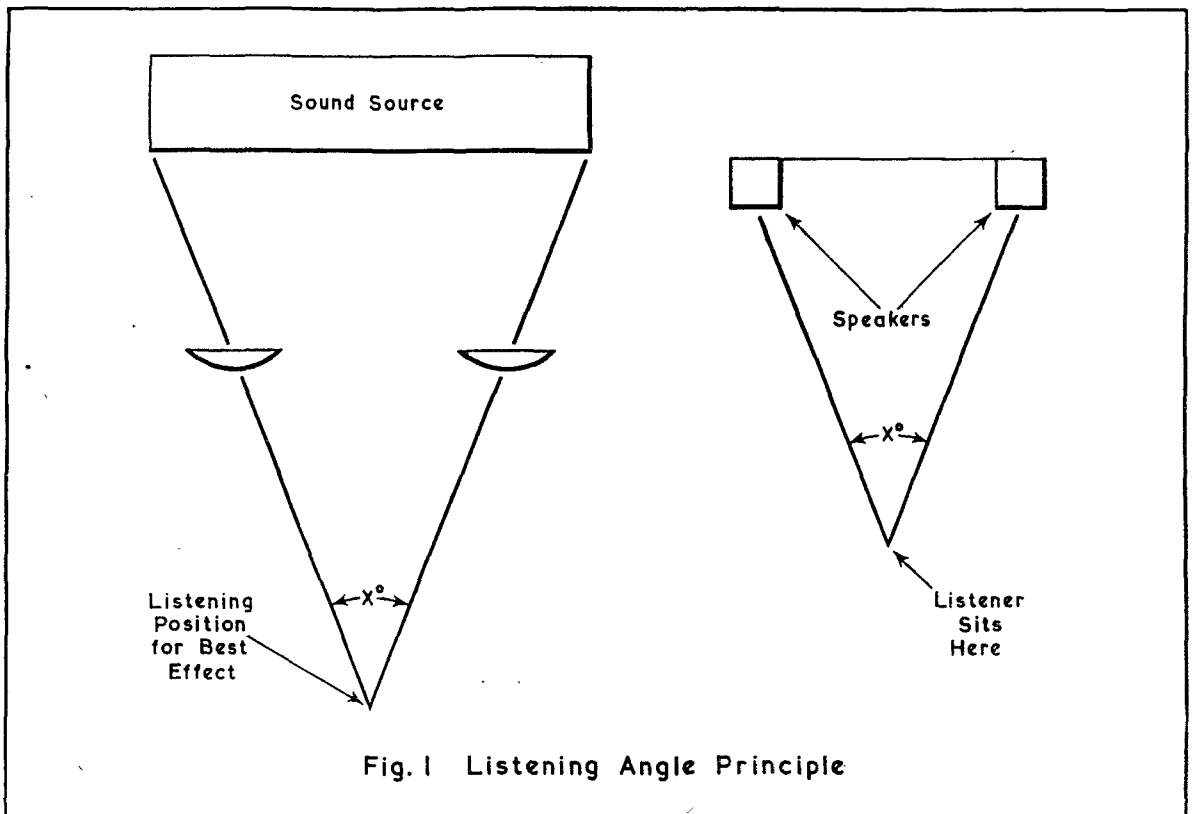
another, sound often appears to come from the centre instead of having wide spacial distribution. Improper phasing can also lead to partial cancellation of some frequencies due to one speaker's diaphragm moving in while the other is moving out.

Use of Microphones

1. **Classical Stereo Recording.** In this system the microphones are placed at the left and right of a parallel line to the sound source. The microphones are usually spaced between 6 and 20 feet apart, sometimes more in order to enhance the effect of spacial distribution. For Binaural reproduction, i.e., through earphones, the microphones are usually placed about 6 inches apart. The object between them simulating the human head. Sometimes, although infrequently, the last technique is employed for stereophonic purposes. The frequencies where the stereophonic sound effect is most pronounced, namely about 1,000 cycles there is substantial

phase differences in the sounds reaching each of the two closely spaced microphones, hence even though the speakers used in reproduction are several feet apart there can be some kind of stereophonic effect resulting from microphones only six inches apart. When microphones are spaced a substantial distance often a centre microphone is also employed, at some stage in the recording process sound from the centre channel is added to left and right.

2. **Listening Angle Principal.** This is sometimes employed in left right recording. The microphones at the left and right as shown in Fig. 1 are spaced so that they are on the angle formed between a listener in a favourable seat at the original performance at approximately the extreme ends of the music source. It is intended that the same angle should be formed between the listener and his two speaker systems, hence the microphones and speakers, through a common angle, in effect attempt to



put the listener in a favourable seat he might have occupied at the original performance.

3. Longitudinal Recording. In this system the microphones are placed along a line at right angles to the music source, i.e., from front to back. This results in a time delay between channels as well as differences in the amount of reverberation. For example, the microphone close to the source picks up more direct and less reverberated sound. Reverberation merely means the effect produced by reflections of the sound from the various walls in which the sound is recorded.

4. Mid-sight Recording. This employs one Cardioid microphone and one Cosine microphone very close together. The Cardioid is orientated to pick up all the audio information which may be called $a+b+c$ with 'a' representing the left, 'b' the right and 'c' the

centre. The Cosine microphone is placed so its figure 8 reception pattern is parallel to the sound source thereby picking up more of the sound on the left (a) than on the right (b) and in the centre (c). The (a) sound picked up by the Cosine microphone is 180° out of phase with the (b) sound inasmuch as the microphone has but one pressure sensitive element which obviously cannot move two ways at once. Hence the sound picked up by the Cosine microphone may be called $a-b$. Fig. 2 shows how the signals of the two microphones are combined. The $a-b$ signal plus the $a+b+c$ signal produces a $2a+b$ signal. The $a-b$ signal is then combined out of phase thus becoming $b-a$ with the $a+b+c$ signal producing a $2b+c$ signal. One channel contains information principally from the left and the other contains information principally from the right, each also contains some centre information.

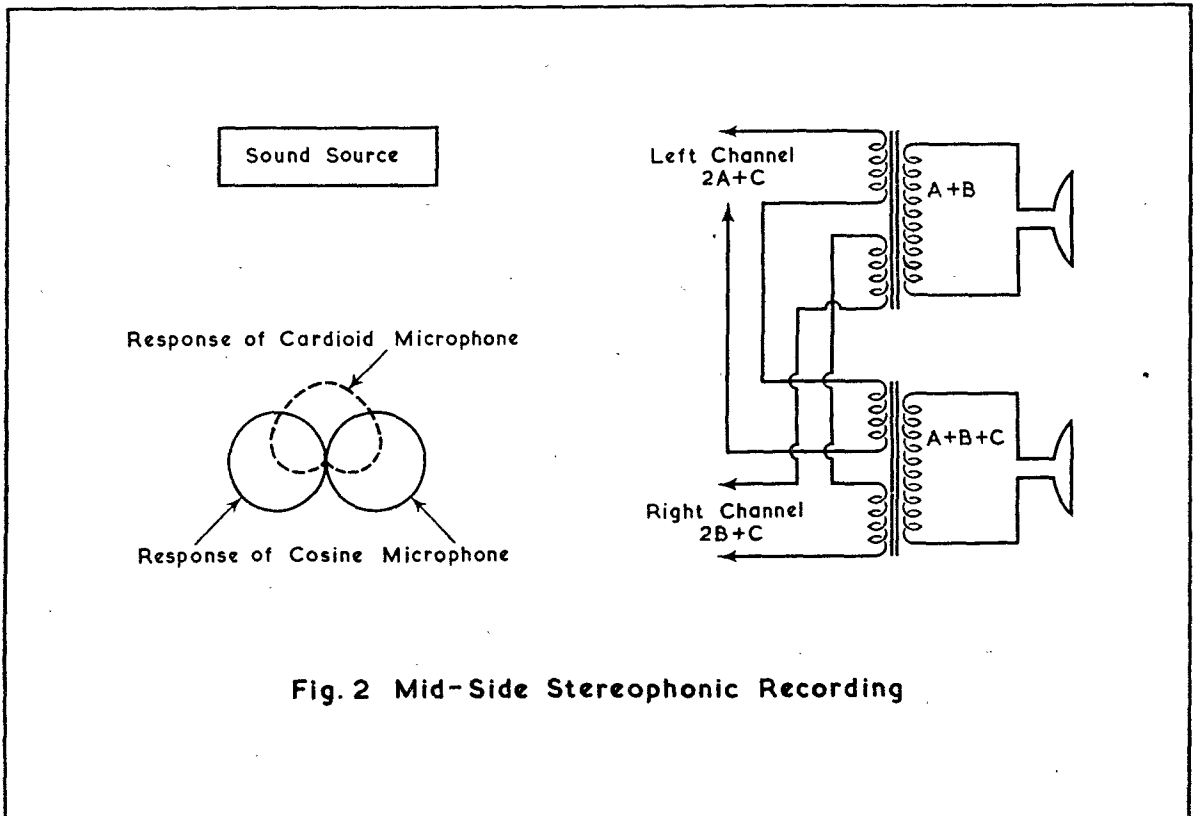


Fig. 2 Mid-Side Stereophonic Recording

Stereo Problems

1. **Hole in the Centre Effect.** Sometimes, if the microphones or stereo speakers are placed too far apart, there is an apparent absence or insufficiency of sound in the centre of the two speakers. In the case of an orchestral composition it might seem that the right and left halves of the orchestra have been sundered and moved a considerable distance apart.

2. **Dummy Speaker.** A psychological device to overcome the hole in the centre effect, consisting of a speaker system or merely a speaker enclosure placed between the left and the right speakers. Although no signal is fed to the centre speaker nevertheless for some persons the visual presence of the little speaker helps to create the aural illusion of sound coming from the centre.

3. **Cross-talk.** Undesired reproduction on one channel of audio information intended for the other channel, occurs to a slight extent in Inline heads where magnetic coupling causes the upper head to pick up from the lower head some of the signal which the latter has picked up from the lower track of the tape. The lower head picks up, of course, the upper track signal in a similar fashion. Cross-talk is sufficiently low in modern stereo heads to be a negligible problem for stereo purposes, the undesired signal being 40 decibels below or more. However, if one half of a stereo head is also used for playback on two-track monaural tape, cross-talk may be annoying, depending on the quality of the head. The best heads keep cross-talk to inaudible proportions. Cross-talk is of greater magnitude on stereo discs where the undesired signal may be only 20 decibels down.

CHAPTER II

POSITIONING THE LOUDSPEAKERS

Perhaps the most difficult of all the problems encountered in stereophonic sound is the placement of loudspeakers. Because of the size and acoustic properties of a concert hall or broadcast studio where musical programmes are recorded on tape it is not difficult to understand that it is almost impossible to duplicate the same acoustical conditions in a normal room. There is no hard and fast system of placing the loudspeakers and it can never be considered as an exact science however there are certain points which must be taken into consideration if the results are even to approach the desired effect.

The final analysis will always lie with the listener who has to determine his own particular preference between the several possible speaker positions.

When a dramatic demonstration of stereophonic sound is required, the ideal positioning of the loudspeakers is as far apart as possible along one wall of the room facing into the room at a slight angle. The current motion of the sound from one corner of the room to the other will then be very obvious. This type of speaker placement in rooms with hard acoustical surrounding, i.e., without carpets and curtains and wall draperies allows enough diffusion of the sound from reflection from walls to ceiling and from floors to cause great loss of the stereophonic effect.

Placing the speakers too close together may be equally ineffectual, for no matter at what angle you set the speakers most of the effect of the stereophonic reproduction will be lost. In general it has been found that a spacement of between 4 ft. and 6 ft. tends to give the

most realistic results. For finest results, however, one must experiment with the actual equipment and judge for oneself which spacing and arrangement produces the nearest approach to perfection. Suggested speaker arrangements for different types of rooms are dealt with separately in this chapter. Fig. 3 shows a typical rectangular room without taking into account the positions of chairs, couches, windows, curtains, carpets, etc. Taking the short side of the room to be of length x it has been found by experiment that about the ideal separation of the speakers is $.7x$. The speakers being symmetrically positioned about the centre line of the room. The best listening area will then also lie equally on either side of the room and will consist of an area of approximately $\frac{2}{3}$ rd x by $\frac{1}{3}$ rd x separated by an average distance of x from the centre of the two loudspeakers. This sort of situation is of course vastly over simplified

because it is not often that one wants to listen to stereo in a completely empty rectangular room. The problems created by such items as sofas, tables, armchairs, curtains, and doorways cannot be overlooked.

Fig. 4 shows a typical drawing room in which the seating arrangement has been set out along one of the long walls of the room. Those unaccustomed to the complexities of stereophonic reproduction often position their loudspeakers at points A and B marked on the diagram, the result is much the same as if two separate orchestras were playing in opposite corners of the room, all semblance of sound distribution is lost. The result being a large hole in the middle effect.

The situation may however be considerably improved by positioning the speakers as

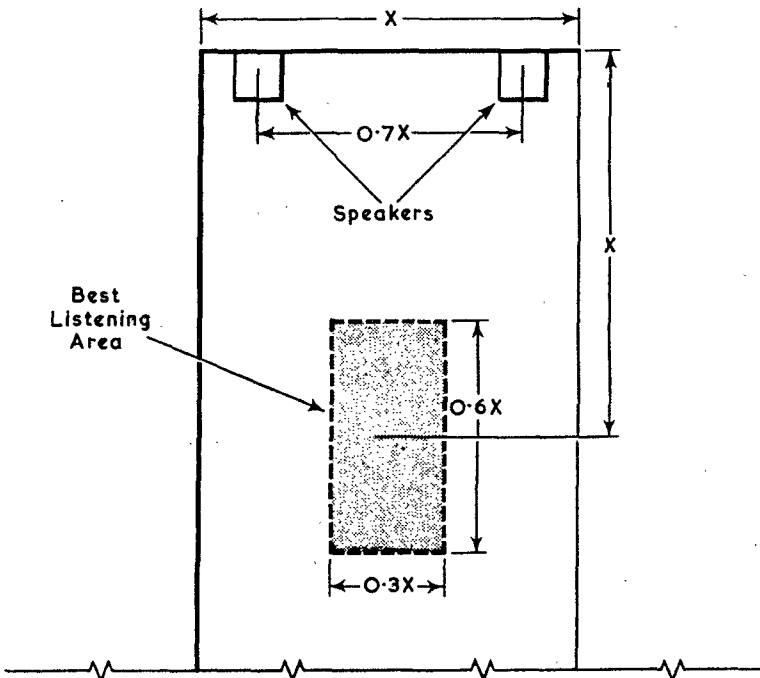


Fig. 3

shown in the diagram, here the length of the room is taken to be x . The speakers are placed symmetrically about the open doorway each being a distance of $\frac{1}{4}x$ from either the centre or the ends of the room. The entire area MNO now receives an admirable stereophonic effect and other points outside this area, separated from the speakers, receive somewhat lessening effect.

Exactly the same room may be arranged in another way, as shown in Fig. 5, where the speakers are placed at angles approximately 30° near one of the short walls. In this case, the stereophonic effect will be best obtained near the centre of the room, for this reason the seating arrangement is moved somewhat from the far end of the room from the loudspeakers. If the chairs were placed along this far wall the stereophonic effect would be to a large extent lost as the different sounds from

the two speakers would, by this time, have merged into one again just as the back row in the theatre obtains very little directional effect from an orchestra, in other words it is hard to separate various instruments except by means of differentiation of frequencies. The angle at which the two loudspeakers are placed to the long walls may have to be varied somewhat depending upon the ratio of the length of the room to the breadth. The longer the room is the more difficult it is to obtain a true stereophonic effect.

A design which has become more and more popular in recent years with the architect is the 'L' shaped combined living room and dining room which is shown in Fig. 6. This is an extremely difficult room for positioning speakers because it has no true corners, furthermore, these rooms rarely have more than one very long wall and that wall is very

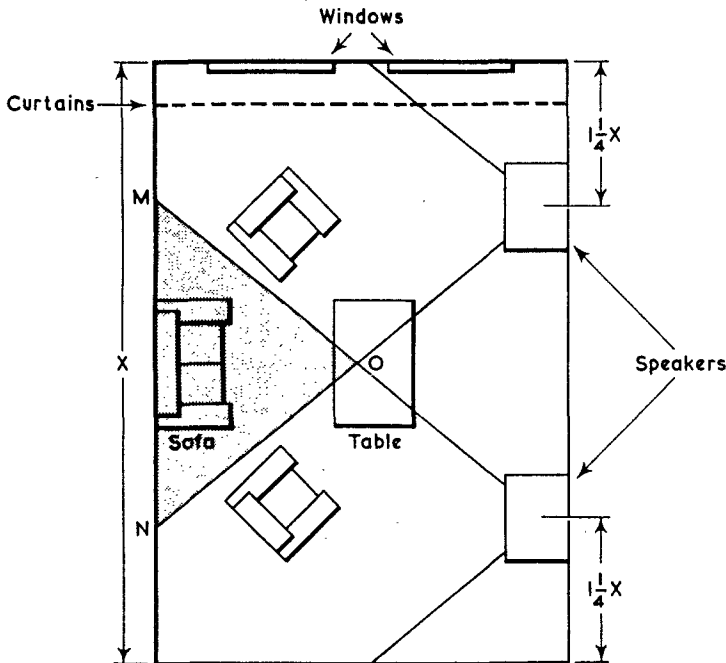


Fig. 4

often too long. This diagram does suggest one solution though it is by no means perfect. The two loudspeakers are placed along the sections of one line of the 'L', each being a distance of $\frac{1}{2}x$ from the end. This immediately introduces the disadvantage of having no backing to the speakers and if a reasonable result is to be obtained it is advisable to put screens, possibly of the folding type, behind the loudspeakers. As these can be extremely attractive they may even enhance the appearance of the room. The ideal listening position now is approximately where the settee is, moving the settee forward might improve the results in certain types of room. The existence of doors

might make this arrangement impossible, however a slight re-arrangement is not likely to upset the stereophonic effect too much.

In certain cases it may be difficult or impossible to sit exactly half-way between the two speakers. This is not absolutely essential, what is essential is that the levels of volume arising from the two speakers is substantially the same. If one is sitting further from one speaker than from the other it is advisable to have a balance control on a long lead taken, say, to the arm of the armchair in which one is sitting, it is then possible to adjust the speakers until their sound levels are apparently equal.

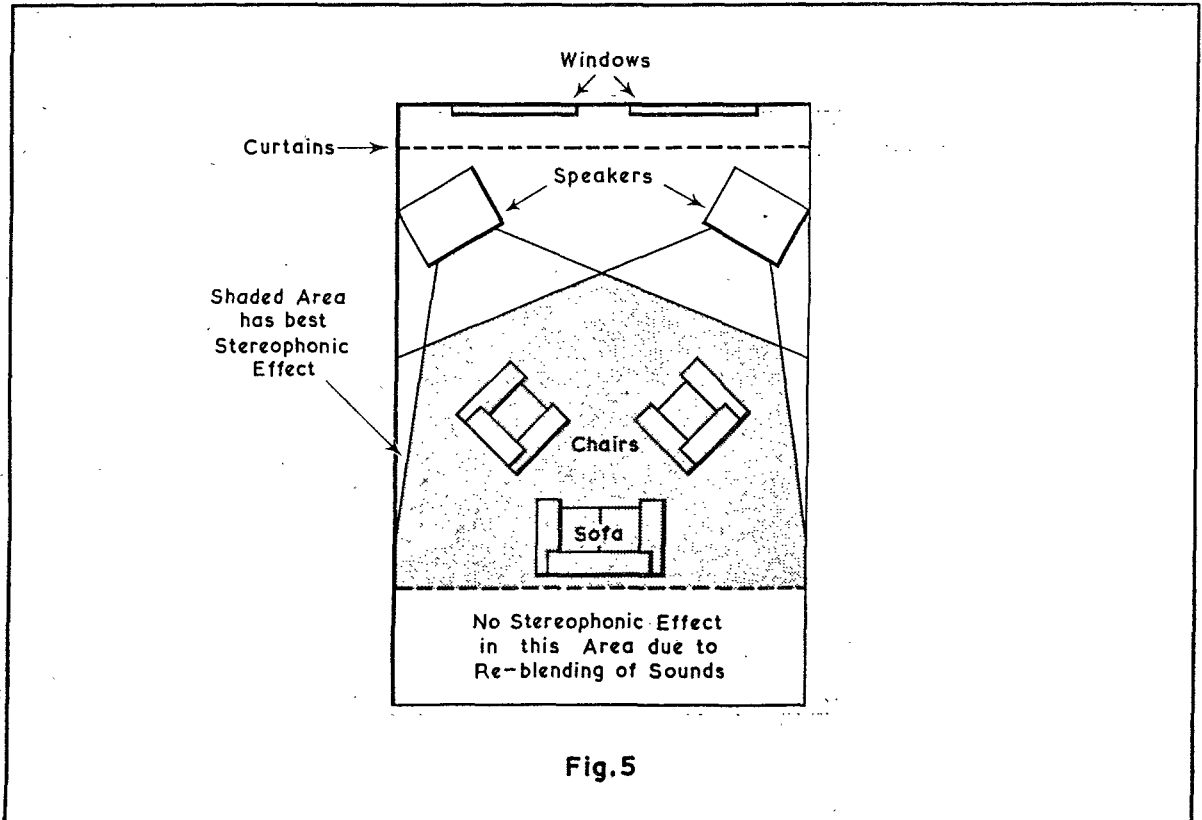


Fig.5

CHAPTER III

HEADPHONES FOR STEREO

The main reason why stereophonic reproduction can never be anything but a good impression when using speakers, is that each ear always hears a large percentage of the information intended purely for the other ear. Thus, although each microphone picks up only the information intended for it each ear still hears both sides of the original sound.

There is, however, a very simple solution to this problem—use headphones. The results are almost unbelievable and are so incredibly better than those produced with loudspeakers that it is surprising that they are used so rarely. As well as the vastly improved performance possible there is another important advantage. The power required by the head-

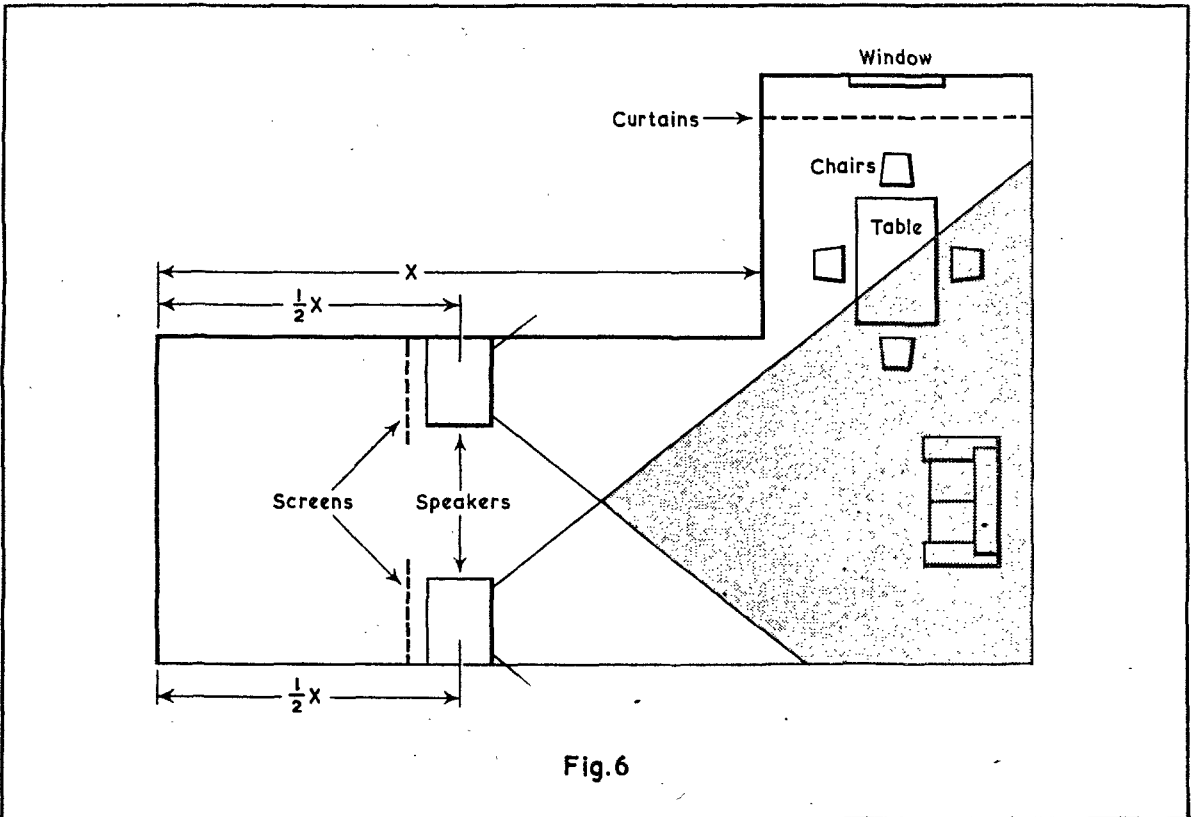


Fig. 6

phones is, at the most, a matter of only a few milliwatts, this makes the job of designing the amplifier very simple and the actual cost is also reduced very considerably.

It is well worthwhile investing in a pair of first class headphones such as S. G. Brown's 'K type' which sell for £5 10s. and which are available specially wired for stereo. Their frequency response is from 20 to 12,000 c.p.s. which makes the reproduction really high-fidelity. To purchase two loudspeaker sets with similar response would cost something like £40. The saving in the cost of the amplifiers is very similar. A complete stereo amplifier similar to that shown in Circuit 1, will cost only about 30s. and as only a single stage is used for each channel, the fidelity is high and the noise level low.

As the output required by the headphones is so low a transistor amplifier offers several advantages and several varieties of this follow. Although the amplifiers are shown powered by batteries in the diagrams there is no reason why a small power pack should not be made to enable them to run off the mains.

CIRCUIT 1

This little amplifier is probably the simplest stereo amplifier that can be made. Each channel uses one half of the Mullard ECC83 double triode. The auxiliary diagrams show how volume and tone controls may be added if desired. Details of the chassis are also shown and it is worthwhile following them though by no means necessary.

For those who are interested in the technicalities, the valves are used as cathode followers. At first sight, then, it would appear that they provide a loss rather than a gain since the voltage gain of a cathode follower must always be slightly less than one. The valves, in this case, act as transformers matching the high impedance of the crystal pick-up to the low impedance of the headphones. Thus, if the impedance of the pick-up is 1 meg. ohm and that of the headphones is 50 ohms each valve has the same effect as a transformer having a turns ratio of about 140 to 1, but with this difference. Although the transformer gives a

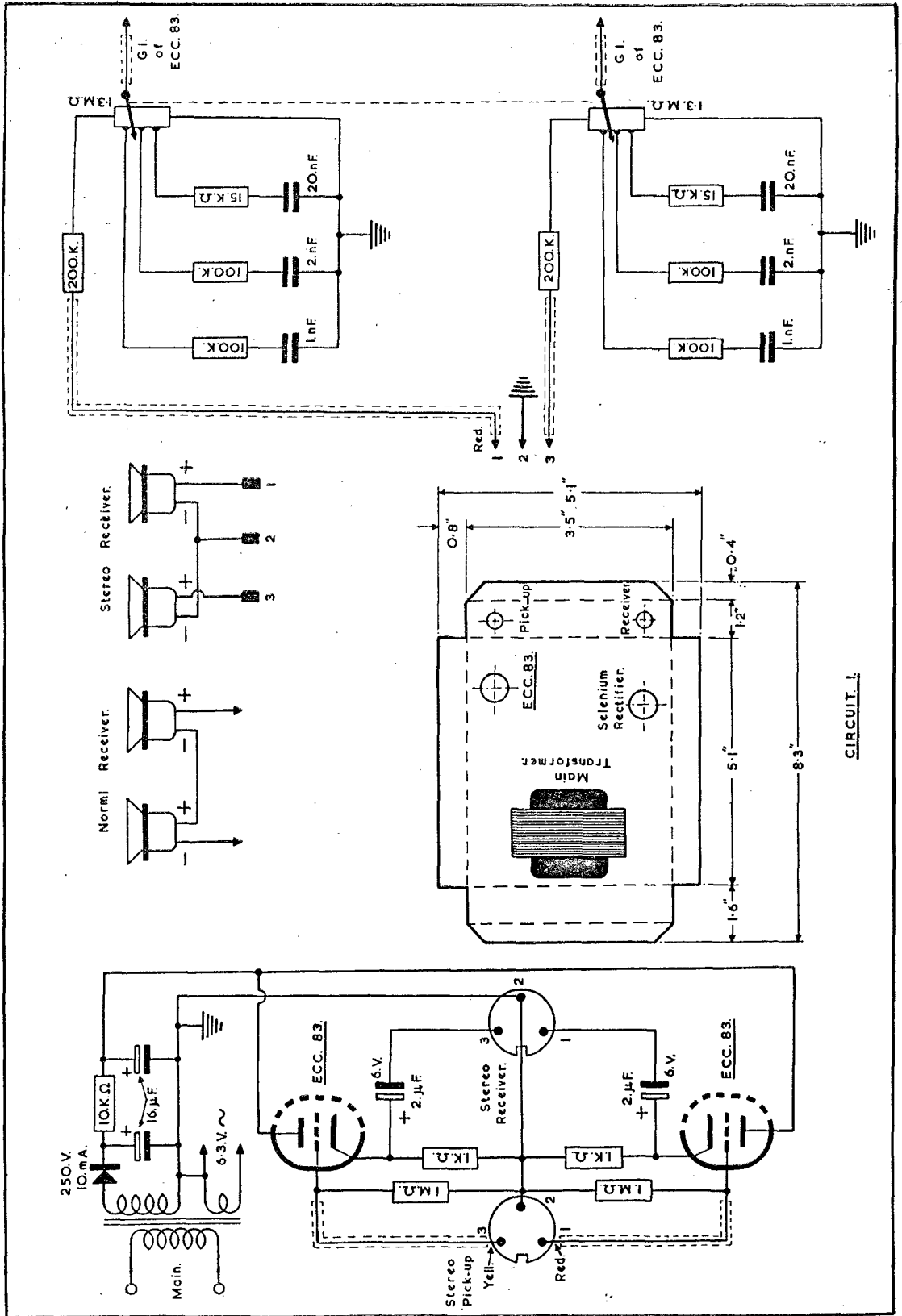
current gain of 140 it also gives a similar voltage loss whereas the valve gives hardly any voltage loss at all. The overall effect, then, is that of a power gain of about 135 times or 21db.

CIRCUIT 2

The second circuit shown here performs exactly the same job as that shown in Circuit 1. Instead of using valves however, it uses transistors with the following advantages. As no heaters are used there is no possibility of hum being introduced and with headphones the slightest hum is immediately noticeable.

The transistors shown are Mullard OC71 but almost any type of small signal PNP transistor would be equally effective. If NPN transistors are used then the battery polarities must be reversed. The input as shown is for use with high impedance pick-ups such as the crystal type. As the input impedance of the transistor is comparatively very low, the signal is fed through a 470 K ohms resistance R1. This effectively matches the transistor input to the pick-up output. Because of the transistors low input impedances all the condensers associated with coupling and de-coupling have to be relatively very large. However, as the voltages involved are very small they need not be very large physically. R3, R5 and C2 are used to stabilise the bias on the base of the transistor. This is important as it reduces the variation in the operating bias condition due to temperature variations.

Unlike the valve, even with no bias the transistor still has a certain amount of conduction due to leakage from the collector to the base. As the temperature rises this leakage increases thereby causing the collector to draw more current, which again increases the temperature of the transistor. This may be guarded against as it is in this circuit by feeding back the D.C. component of the output so as to be out of phase with the input. In the case of the junction transistor the output is automatically out of phase with the input and by putting the resistance in the emitter lead, the necessary feed back is achieved. However, normally, this feed back would be equally



CIRCUIT. I

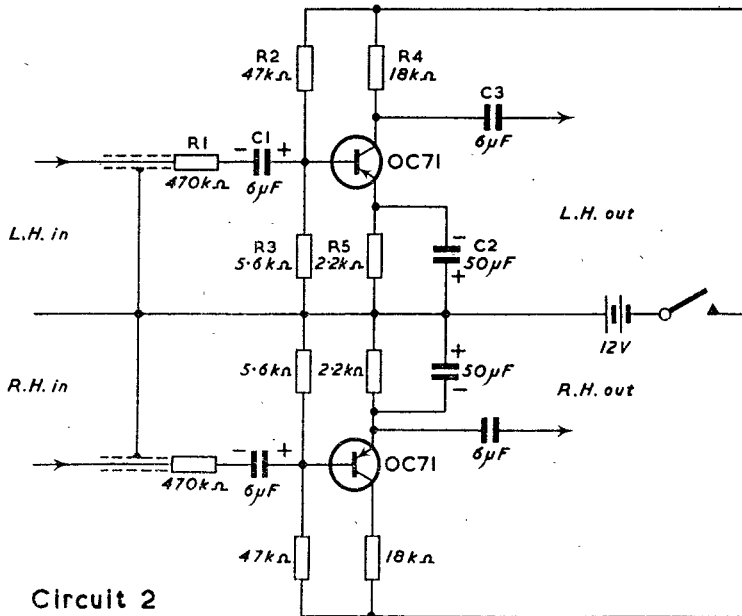
effective to D.C. and A.C. and the signal gain of the transistor would be very low. To avoid this, R5 is by-passed by a high value electrolytic C2, 50 microfarads. The output is taken via two 6 microfarads condensers to the headphones. These should have as high an impedance as is possible, preferably at least 4 K ohms for each headphone. Normal headphones when purchased are wired in series, for stereo they must be re-wired so that one side of the headphone is taken to one amplifier and the other side of the headphone to the other amplifier.

Under normal conditions a typical gain for an OC71 in the common emitter configuration would be approximately 20 decibels, however due to the losses involved in a 470 K ohm matching resistance, this gain will be markedly reduced and in the case of some low output pick-ups, will be insufficient to enable the transistor to drive the headphones to sufficient volume, in which case a second similar stage

may be added or one of the circuits later on in the book may be used. Construction of this circuit is in no way critical but the input and output leads should be kept separate as far as is possible.

Newcomers to transistors should remember that in some ways they are rather more delicate than valves. For example, they are sensitive to heat and if they are to be soldered into the circuit, a heat shunt should be used to hold the leads. If transistor sockets are used the transistors must not be plugged into these sockets with the battery switched on as this may ruin them. Another way in which the transistors may be easily ruined, if care is not taken, is by reversal of battery and the circuit should always be checked before switching on for the first time.

Considerable emphasis has been placed on transistor amplifiers in this book, and some readers may feel that an insufficient number of valve amplifiers has been given. However,



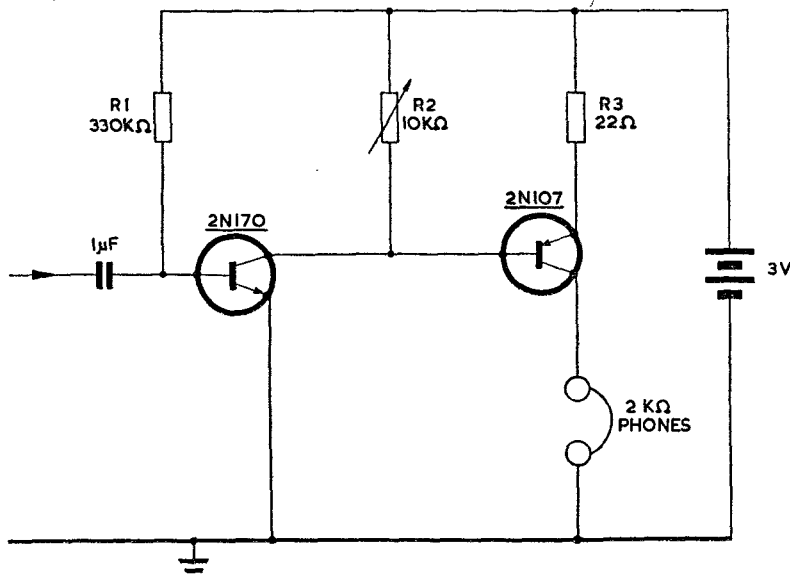
transistors are very rapidly replacing valves in all audio circuits and the time will come when no more valve amplifiers will be produced. The prices of transistors are rapidly dropping and in a few years will definitely be lower than valves. It is already possible to buy some audio transistors for as little as 7s. 6d. and eventually they should be readily available at 2s. 6d. a piece or even less.

CIRCUIT 3

As was mentioned when discussing Circuit 2, occasionally a single transistor amplifier provides insufficient gain to drive the headphones. Circuit 3, therefore, shows a two-transistor amplifier. Only one amplifier is shown and for stereo, of course, two must be used but as these will be identical it was not thought necessary to duplicate them. As shown, the amplifier is suitable for use with low impedance pick-ups. If it is required for use with high impedance pick-ups then a 470 K ohm

resistance should be inserted in the input lead.

The transistors shown are of American manufacture and no equivalent is at present available in this country for the 2N170, however, transistors of this type should shortly be available from Technical Suppliers Ltd. (TSL) at a very reasonable price. As may be seen the circuit involves direct-coupling between the first and the second transistor, this has several advantages not the least of which is the marvellous economy of components. The first transistor and the second transistor are symmetrical to one another that is to say, one is NPN and the other PNP, so that they require opposite battery polarities. Base bias for the second transistor, i.e., 2N107, is supplied through the emitter and collector of the first one. If a strong signal is applied to the base of TR1 its collector current will increase and increase the base bias on TR2 automatically. At the same time the collector current for TR1 is partially supplied via TR2. When using two



R1 should be adjusted for optimum results

Circuit. 3.

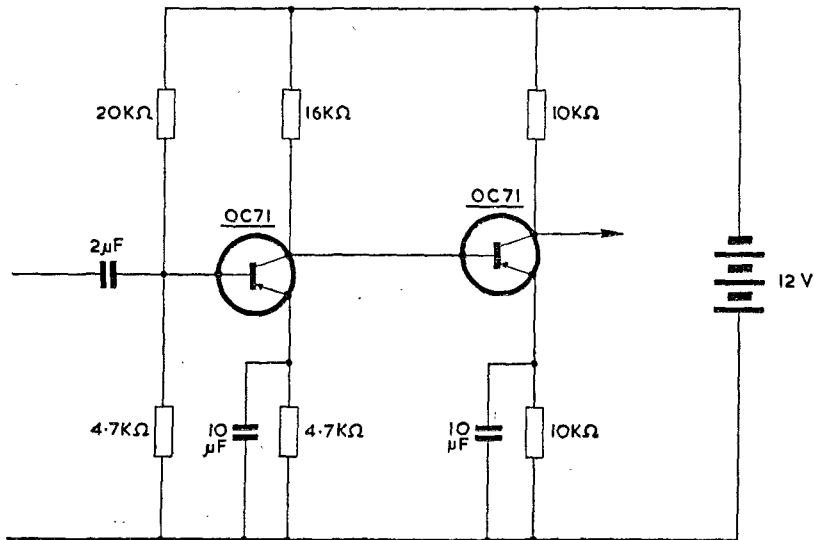
such amplifiers for stereo, the two 10 K ohm resistances should be ganged together so that the volume control operates both amplifiers at the same time. R3 which, as may be seen from the diagram, is in the emitter lead of TR2, is not by-passed by an electrolytic condenser as a similar resistance in the last circuit was. This is because a certain amount of negative feed back of audio frequencies is very useful in improving the frequency response of the amplifier. If R3 is omitted an increase in gain will result but at the expense of a certain amount of gain.

CIRCUIT 4

This amplifier is very similar to that shown in the last diagram. Instead of using an NPN and a PNP transistor however, it uses two PNP types which are readily available in this country at the moment. In the diagram a 10 K ohm resistance is shown as the collector load of the last transistor, however, this may be

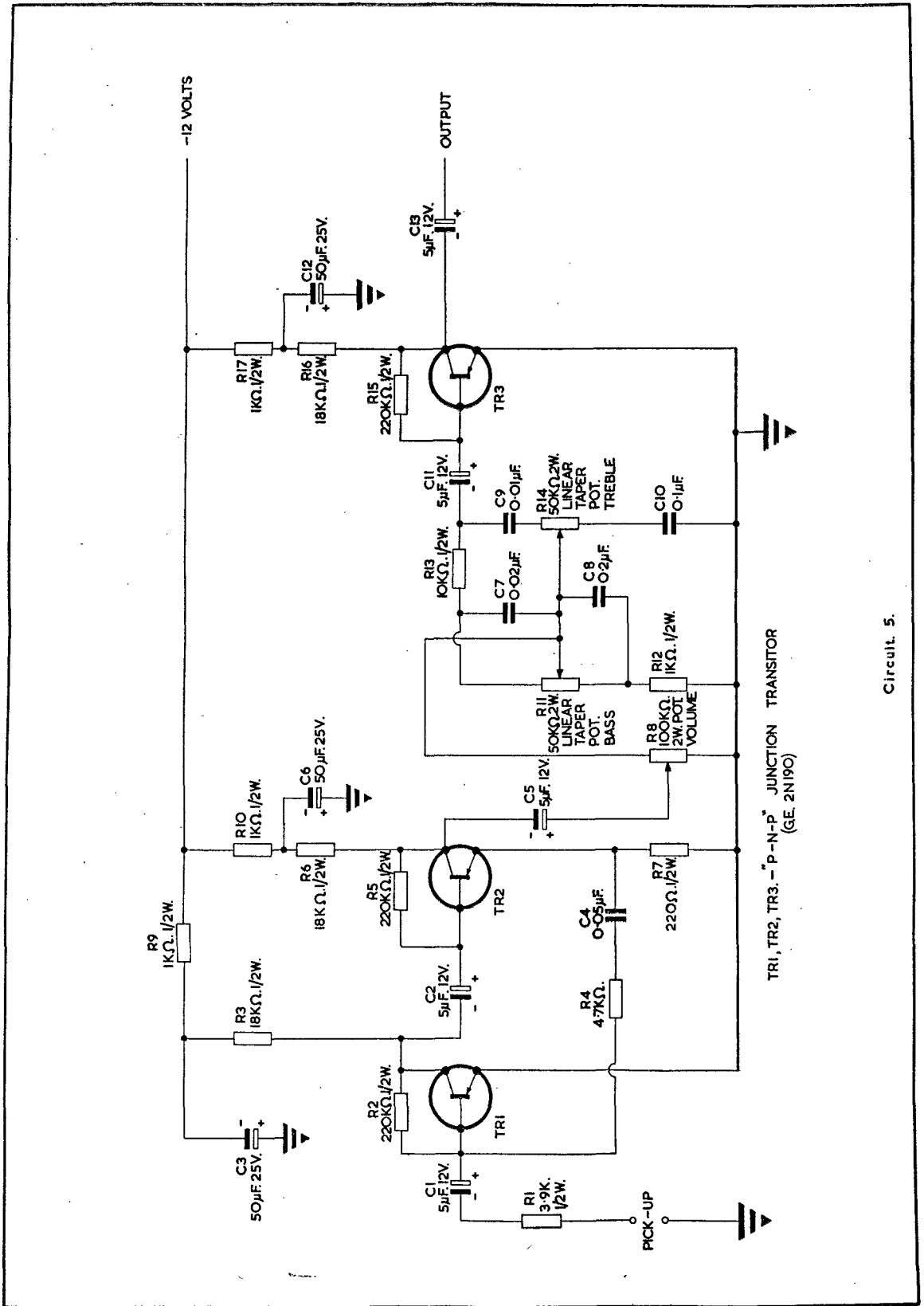
replaced by a headphone, one side of the headphone being used for each stereo channel. The input, as shown, is for low impedance pick-ups, for a high impedance pick-up resistance of approximately 300 K ohms should be placed in the input lead. A little trial and error with this value will determine which resistance gives the best result.

Direct coupling between the two transistors is again used, this time however the method is somewhat different. The collector load for the first transistor is also the resistance which supplies the base bias to the second, i.e., the 16 K ohm resistance. If transistors other than the OC71 are used it may be necessary to alter this resistance as it is somewhat critical. If various transistors are to be tried at different times in the circuit, then a 25 K ohm semi-variable potentiometer may be used instead of the fixed type shown. If this is done, however, care must be taken to ensure that the value of this resistance does not fall below



Direct-Coupled A.C. Amplifier

Circuit. 4.



TR1, TR2, TR3 - "P-N-P" JUNCTION TRANSISTOR (GE 2N190)

Circuit 5.

about 10 K ohms at any time, as if the current for the second transistor becomes too great in the base, the collector will draw more current than this type of transistor should and will be destroyed. In this, as in every battery-operated transistor circuit, a high value electrolytic connected across the battery having a value of say 100 microfarads, will improve the battery life. This is because as the battery gets older, the internal resistance increases and causes coupling between the two transistors, or however many stages there are, which may result in slow relaxation oscillations.

CIRCUIT 5

Circuit 5 shows a three-transistor high-fidelity pre-amplifier with volume, bass and treble controls. By using an un-by-passed resistance in the emitter of the second stage, a voltage is obtained which is proportional to the output current of the amplifier. If a resistance and a capacitor are connected to this resistor, as they are in this circuit, a signal is fed back to the input which is proportional to the output current. If the feed back capacitor is made very large, the frequency response is essentially flat and gain is determined only by the ratio of R4 to R7. If the capacitor is made small the feed back current will depend upon the frequency being amplified and it is possible to obtain a boost of the low frequencies. With the values shown the amplifier provides compensation for a variable reluctance pick-up reproducing from records recorded to the usual standards. In valve pre-amplifiers, feed back voltage is usually obtained from the anode of the second stage and applied

to a resistor in the cathode of the first stage. This method of feed back is not well suited for an all-transistor amplifier since voltage feed back tends to control the voltage applied to the next stage whereas it would be more desirable in transistor amplifiers to control the current in the next stage.

If a transistor pre-amplifier is to be used with a valve amplifier however, voltage feed back can be used successfully. The three transistors all use resistance capacity coupling in the interests of high-fidelity. As may be seen from the diagram the base bias is taken via a resistance direct from the collector, i.e., R2 for TR1, R5 for TR2 and R15 for TR3. As the collector is out of phase with the base this means that negative feed back is always applied at D.C. as well as A.C. thus the transistor is effectively stabilised against changes in the bias which would normally arise due to variations in temperature. The effective input impedance of the pre-amplifier may be varied by varying R1 thus for high impedance pick-ups of the crystal type, a resistance of about 200 K ohms will be necessary, whereas for a very low impedance pick-up no resistance at all need be used. As two pre-amplifiers will be used for stereo the volume, bass and treble controls must each be ganged to their similar components in the other channel. The transistors as shown in this diagram are made by General Electric under type No. 2N190 and it is hoped that TSL will soon have these available in this country. However, a suitable replacement would be the OC70 for TR1 and TR2, and the OC71 for TR3.



CHAPTER IV

LOUDSPEAKER STEREO AMPLIFIERS

In the last chapter, it was pointed out that marvellous high-fidelity results could be obtained from headphones, however, this system is of little use if a large number of people are to listen to the recording, as to supply a large number of headphones of suitable quality is both expensive and inconvenient. If the loudspeakers are arranged properly, according to the instructions given in Chapter II, the stereophonic effect can be really excellent. However, haphazard arrangement of loudspeakers can produce very poor results, so considerable care should be taken and a little experiment will pay large dividends.

Some households may find difficulty in finding space for two loudspeakers in the living room without overcrowding the people who have to sit there. This problem may be overcome, however, by using speakers in corner cabinets such as the TSL Soundcorner which virtually take up no space at all as they are mounted above the floor level. Furthermore, the TSL Soundcorner has one of the finest frequency responses available from any speaker system in this country and at the same time is very reasonably priced. It is therefore admirably suited to stereophonic units.

CIRCUIT 6

This amplifier was designed by Mullard Ltd. at their Applications Research Laboratory and we are very grateful to them for permission to print it in this book. Each channel uses one ECL82 valve, the triode section acting as a voltage amplifier and driving the pentode section which forms the output stage. Negative

feed back is taken in from the secondary winding of the output transformer to the cathode circuit of the triode section. The EZ80 is used in a conventional power supply with resistance-capacitance smoothing. The sensitivity of the circuit is sufficient for use with all existing stereophonic crystal pick-up heads. In addition to the straight forward stereophonic arrangement the compatibility switch enables the outputs from the pick-up elements to be connected either in parallel or in series, depending on the type of pick-up head used for reproduction from monaural recording. The output power available from each channel is 2 watts giving a total output power of 4 watts from the two speakers. The total harmonic distortion is sufficiently low, being only 5 per cent., and sensitivity is 280 millivolts. The frequency response of this amplifier is completely beyond reproach being $\pm 1\frac{1}{2}$ db between 40 cycles and 40 kc/s. The tone control gives a treble cut of 22 decibels at 10 kc/s and the hum and noise are a full 60 decibels below the rated output. The output transformer should be the usual pentode output type to match the output from the ECL82 to a 3 ohm loudspeaker. When building this amplifier the two output transformers should be mounted at right angles to eliminate the possibility of cross-talk between the two channels due to mutual inductance. Amplifiers of both channels should be supplied with the same power supply. The power transformer has a secondary winding of 250-0-250 volts centre tapped, 6.3 volts untapped and 16.3 volts centre tapped. Dual controls should, of course, be ganged together.

CIRCUIT 7

A rather more powerful amplifier is shown in Circuit 7. This has an output power in the region of 5 watts per channel, giving a total power of 10 watts from the two speakers. The output transformers should be of the centre tapped type normally used in push pull circuits. About 8 decibels of voltage feed back is applied from the secondary of each output transformer to the cathode of the corresponding ECC83. Resistances R4 and R3 make up the voltage feed back divider for channel 1, while R11 and R10 perform the same function for channel 2. 8 dB feed back is the maximum amount that can be used with the inexpensive transformers and the limited gain available in the two amplification stages. In the power supply circuit, it will be noted that R1 is made variable this is to enable precise setting of the high voltage for maximum power output with minimum distortion.

CIRCUIT 8

This amplifier, which is ideal for use in small living rooms, has an output of 200 milliwatts per channel giving a total output in the region of 0.5 watt is another developed by Mullard at their Applications Research Laboratory. The amplifier was designed with four Mullard PNP junction transistors, it consists of an OC71 input stage, an OC71 driver stage, feeding a push pull output stage using two OC72s. The power is supplied by a single 6 volt battery. The first and second stages are resistance capacity coupled and the driver stage is transformer coupled to the Class B output stage. The conventional method of D.C. stabilisation is used in all stages. This method of stabilising the D.C. operating conditions is not however used in the output stage. Its use would mean a loss in sensitivity and reduction in output power, but various biasing arrangements are used to overcome the effect of changes in ambient temperature and battery voltage, as well as production spreads in the transistors.

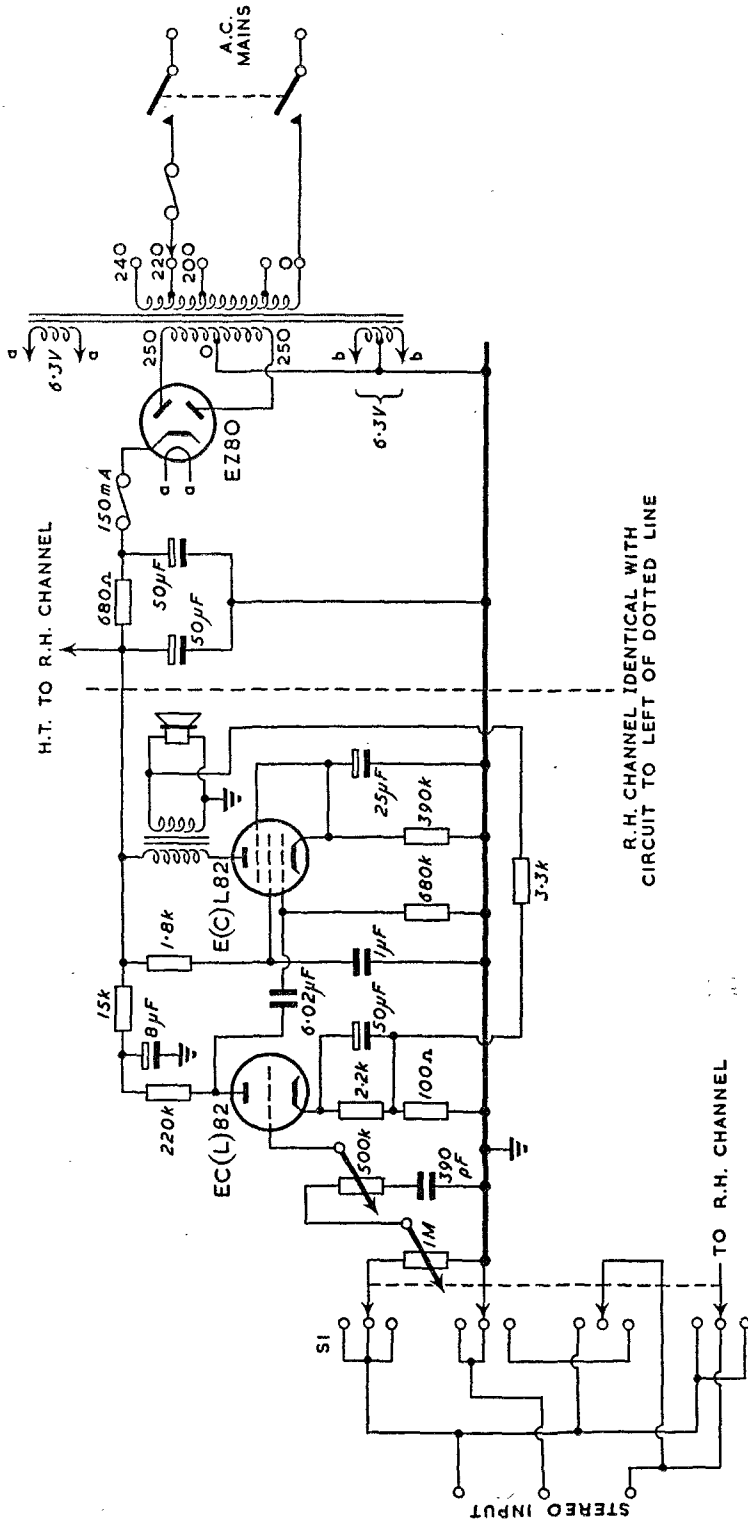
At an output of 200 milliwatts, the total harmonic distortion is in the region of 10 per cent. Frequency response is within ± 1.5 dB

from 50 c/s to 8kc/s. Basic sensitivity at the high impedance input is 400 millivolts and the current drawn from the battery will lie between 5 milliamps at zero drive and 5 milliamps at maximum output with sine wave drive.

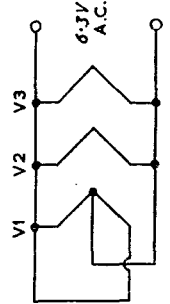
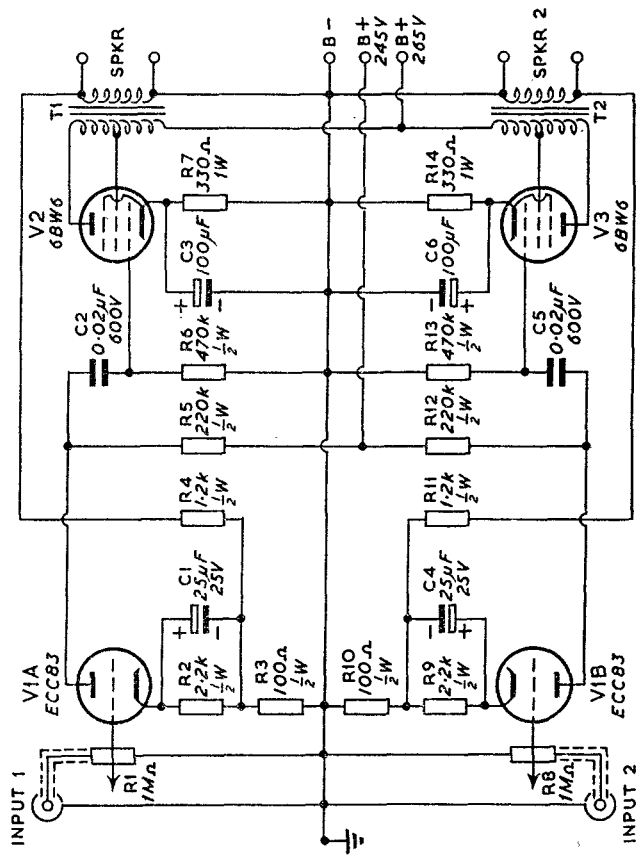
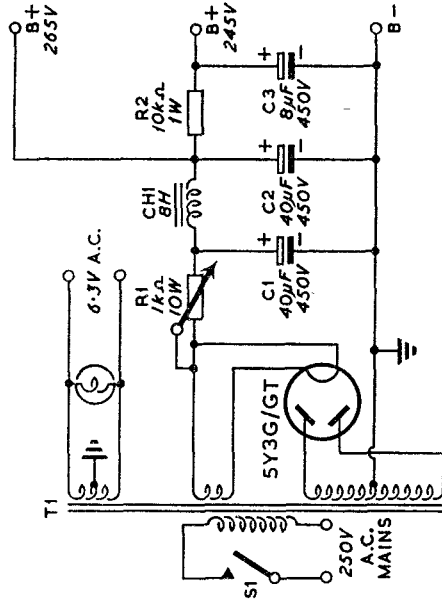
Bias is applied to the bases of the transistors for the first and second stages through the potential dividers R3, R4 and R7, R8 and this arrangement together with the emitter resistors R6 and R9 stabilises the D.C. operating conditions of these stages. Mullard recommend that resistors of 5 per cent. tolerance be used throughout. The driver stage is designed around a collector current of 1.5 milliamps, this was found to give the best operating conditions and permits the inclusion of a stabilising resistor R9 without the required maximum voltage swing being limited. Base bias is obtained from the potential divider R7, R8 and the required collector current is achieved with good stabilisation.

The driver stage is transformer coupled to the output, the resistance of the primary and secondary windings of T1 must be kept as low as possible while the shunt inductance is maintained at a high level. To achieve close coupling between both halves of the secondary, bifilar windings may be used. The turns ratio specified gives sufficient drive for the output stage, together with good sensitivity. A greater step down ratio would give somewhat higher sensitivity but a greater voltage swing would be required from the collector than is available. The shunt inductance of the inter-stage transformer, which is rated at 2 milliwatts, should be 10 Henries with a primary D.C. current of 3 milliamps and the D.C. resistance of the primary windings should be less than 150 ohms. The resistance of each half of the secondary winding in which the D.C. currents are balanced, should be less than 75 ohms and the leakage inductance should be as low as possible.

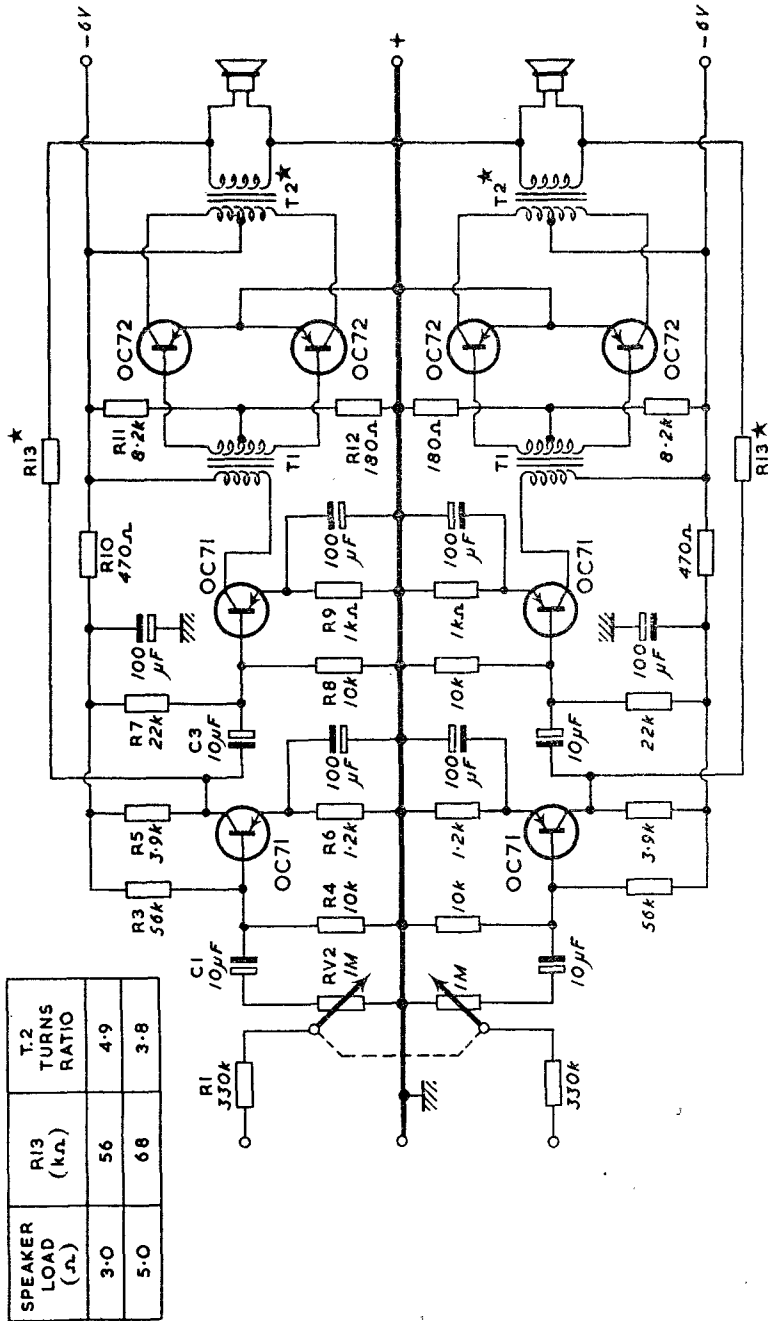
Lower values of shunt inductance will, of course, give a less extended bass response than that quoted. The turns ratio of the inter-stage transformer should be 3.5:1 + 1. As was mentioned in an earlier circuit an increase in the



Circuit 6

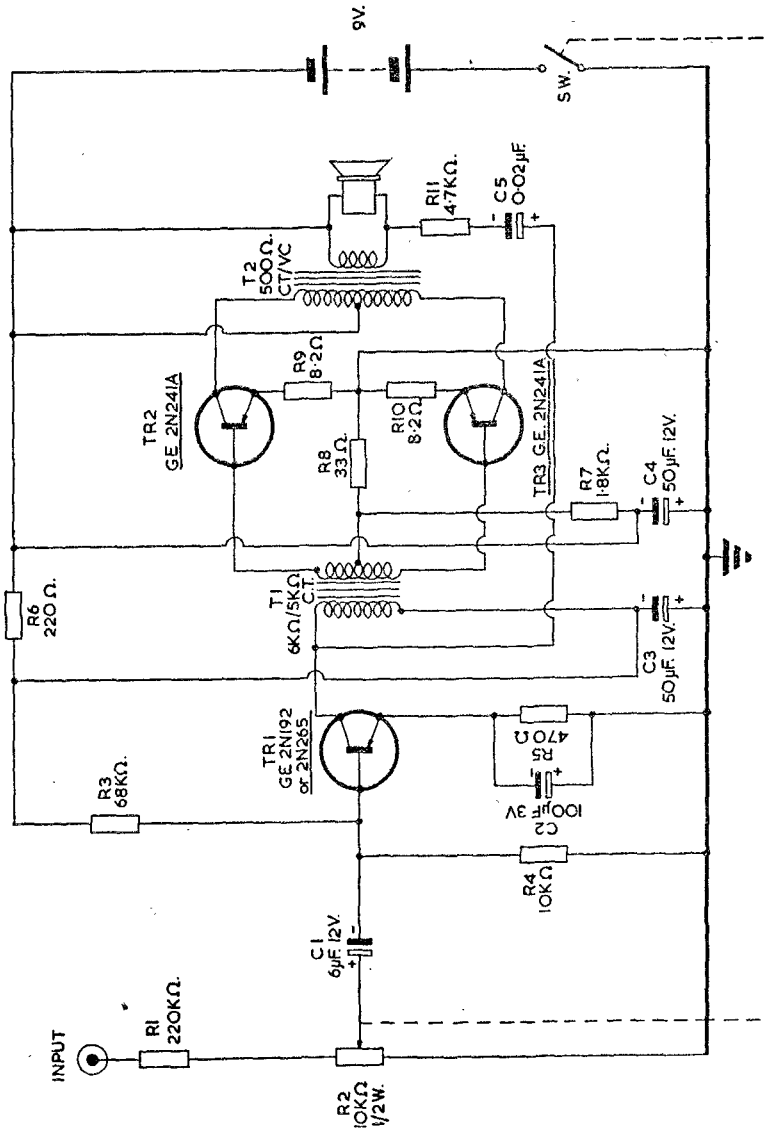


Circuit 7



Circuit 8

★



Circuit 9

battery resistance due to ageing causes undesirable feed back through the supply to the earlier stages, and will result in considerable distortion of the output. This distortion is caused mainly by feed back to the base of the driver, feed back to the collector having a negligible effect. It is possible therefore to introduce an efficient filter circuit to combat this undesired effect at a comparatively low current point and in this circuit it has been added between the collector and base feeds to the driver stage R10 C4.

When the output stage operates under true class B conditions cross over distortion occurs and on sine wave drive it is easily detected at the loudspeaker. If this stage is biased for currents of about 1 milliamp per transistor, this distortion is reduced considerably. For speech and music, however, this distortion is less discernable and the circuit has been designed to give average quiescent currents of between .5 and .6 milliamps for temperatures at which the amplifier is most likely to be operated.

CIRCUIT 9

This circuit is similar to the last but uses three transistors per channel instead of four. The transistors shown are of American manufacture and a possible British equivalent is the Mullard OC72 in each case. The maximum possible power output is 350 watts or 250 milliwatts for 10 per cent total harmonic distortion. Sensitivity for 50 milliwatts reference power output is 200 millivolts. When the circuit is used with a magnetic cartridge R1 should be omitted, in this condition the sensitivity will be 5 millivolts.

The first stage is stabilised in the usual fashion, R3 and R4 supplying the base bias, and R5 and C2 feeding back the D.C. component. C3 and C4 remove the possibility of a high impedance power supply as the battery ages. The driver transformer should have a primary impedance of 6 K ohms and a secondary impedance of 5 K ohms centre tapped. The output transformer should have centre tapped primary 500 ohms impedance and a secondary to match a 3 ohm loudspeaker.

This amplifier like the last is really only suitable in the smaller sized living room due to its limited output power.

CIRCUIT 10

A more powerful amplifier, still using transistors, is shown in Circuit 10. Here the maximum power is 750 milliwatts or 450 milliwatts for 10 per cent total harmonic distortion. With an output power of 100 milliwatts, the maximum distortion is in the region of 5 per cent., the frequency response is ± 5 dB from 50 c/s to 8 kc/s. The sensitivity for 50 milliwatts power output is 120 millivolts at the high impedance terminals and 2 millivolts at the low impedance terminals. The driver transformer T1 should have a primary impedance of 4 K ohms and a secondary impedance of 2.6 K ohms, the secondary being centre tapped. The output transformer should have a primary impedance of 200 ohms centre tapped and the output to match the voice coil of the loudspeaker.

Both sides of the amplifier may be operated from the same power supply, that is the 9 volt battery. In the interests of long life and constant voltage it is well worthwhile using Mallory mercury cells. As these have a voltage of approximately 1.3 volts per cell, seven cells will be required in series to provide the required potential.

CIRCUIT 11

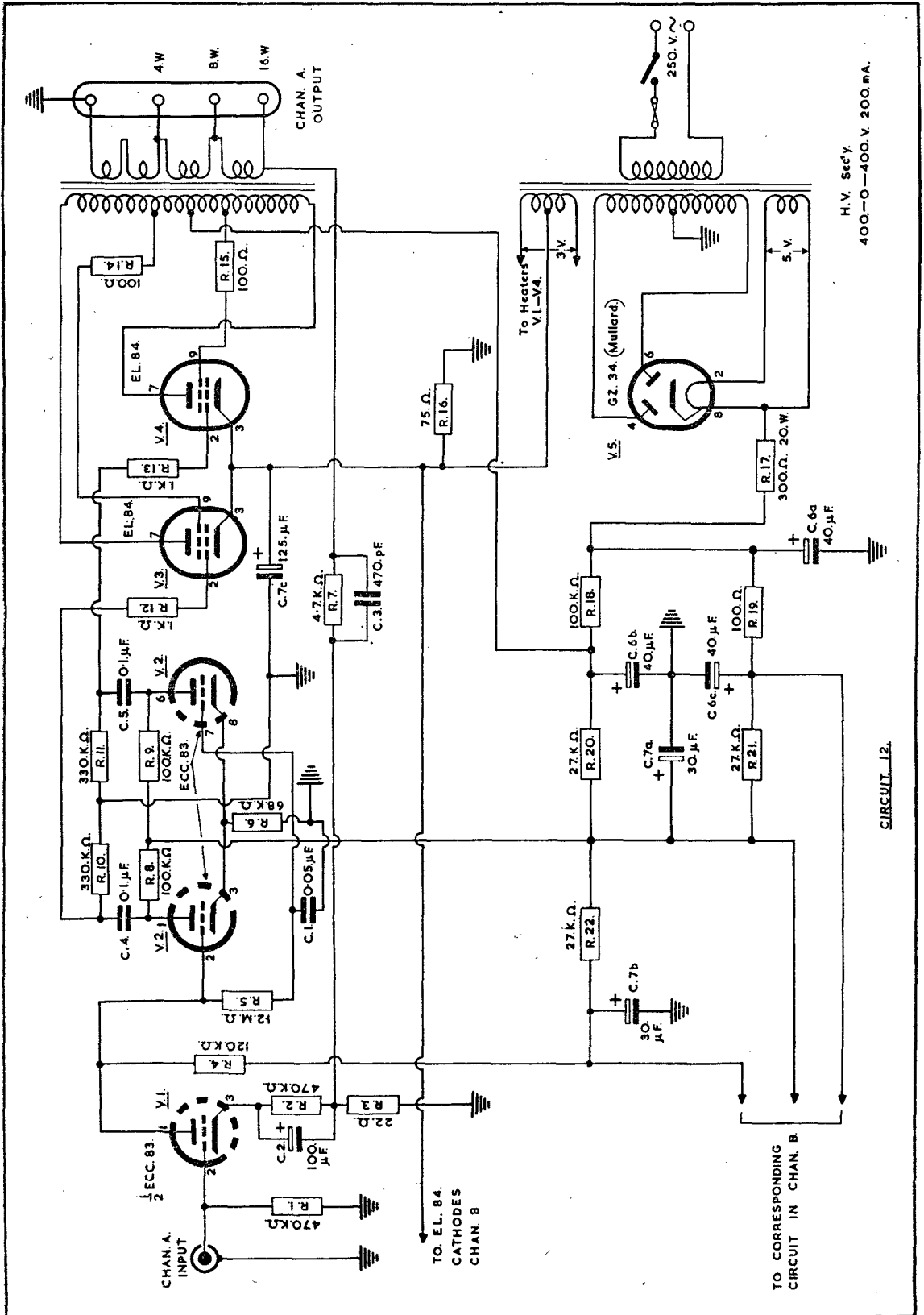
This amplifier is considerably more ambitious than the transistor ones preceding it. TR1 and TR2 are in the form of a compensating pre-amplifier which compensates the irregular response of the pick-up. The resistance and condenser in series from the emitter of TR2 to the base of TR1 achieve this—TR1 is stabilised by the resistance from its collector to its base which also supplies the base bias—TR2 is similarly stabilised. The output from TR2 is coupled to the main amplifier via a 5 microfarad electrolytic condenser and a 100 K ohm volume control. Between the pre-amplifier and the main amplifier are tone control network giving both bass boost or treble boost, by means of the 250 K ohm potentiometers. Variation of the boost given may be

achieved by adjusting the values of the condensers concerned, i.e., .02 and .2 microfarad condensers for the bass and .008 and .08 microfarad condensers for the treble. TR3 is again biased by means of a resistance from its collector to its base and also has a certain amount of negative feed back via the 5 meg ohm resistance from the collector of TR5. The output from TR3 is resistance capacity coupled to the base TR4. TR4 and TR5 are directly coupled together and TR4 is stabilised by taking its bass bias from the emitter of TR5, whilst TR5 is stabilised by the un-by-passed 500 ohm resistance in its emitter lead, which also provides negative feed back at audio frequencies. The output from TR5 is transformer coupled to the power output stage which consists of TR6 and TR7 in push pull coupled by means of transformer T2 to an 8 ohm loudspeaker. T1 should have a primary impedance of 5 K ohms and a tapped secondary of 3 K ohms. T2 should have a centre tapped primary of 125 ohms to match secondary of 8 ohms. In the original design TR1, TR2, TR3, TR4 and TR5 were all General Electric 2N190 transistors, a British alternative for these, however, is the Mullard OC72. TR6 and TR7 were originally 2N188A or 2N320 transistors, replacements for these may be Newmarket power transistors types V15/20IP. Using these transistors the maximum power output per channel of 4 watts should be achieved giving a total power of 8 watts for the complete stereo amplifier.

CIRCUIT 12

For high-fidelity reproduction it is generally agreed that the minimum output power from an amplifier should be in the region of 20 watts, however, when two channels are employed in a stereo system, a similar figure would be in the region of 10 watts giving a minimum output power of 20 watts with the two channels combined. The amplifier described here uses two pairs of EL84's to provide 15 watts output for each channel, giving a total of 30 watts for the complete stereo amplifier. Phase splitting stage is cathode coupled with the input signal being

fed direct to one grid at the same D.C. potential as the anode of the first stage, and with a de-coupling resistor to the second grid, which is by-passed for A.C. to ground, yet maintained at the same D.C. potential as the signal grid. The anode currents for both sections of the tube flow through a large cathode resistance and since one grid is maintained at an A.C. ground potential while a signal is introduced to the other grid, plate current variations across this resistance provide an effective drive to the second section. The circuit has low distortion and when carefully constructed so that the capacitance of the two anode circuits are kept to low and equal values, the overall balance is excellent. To ensure good balance in practice anode load resistors must not be exactly equal. For proper operation with the anode voltage selected the grids of the phase splitter should be maintained at a D.C. voltage of approximately 90 and by selection of the plate load resistance value for the first triode, together with bias resistance the voltage drop across the former may be made to provide this value. The direct coupling between these two tubes results in the zero phase shift at low frequencies and contributes to its stability. The output transformer uses the new Partridge P5000 series, with 10 dB of feed back these transformers are rated at 20 watts at 30 c.p.s. for less than 1 per cent. distortion, or 35 watts at 50 c.p.s. Since the tubes are capable of only about 15 watts this was considered more than adequate to ensure good performance. Leakage inductance from primary to secondary specifies Millihenries which is the same from one half of the primary to the other. Shunt inductance of the primary is high and rated at 450 Henries at a 4 watt output at 50 c.p.s. thus good low frequency performance could be expected. The units are constructed with cast aluminium frames and are very rugged in appearance. The secondaries are wound in four separate sections permitting the use of series connection for a total of 16 ohms output. Combinations of series and parallel connections 4 and 8 ohms and with all 4 ohm windings in parallel, the output impedance of 1 ohm. Since both channels are handling



CIRCUIT 12.

roughly the same signal, common power supply will operate perfectly well.

The layout may best be left to the constructor but the four EL84's are best laid alongside the transformers, while the three ECC83's should be in shielded sockets. So as to provide identical physical layout for the phase splitter and output stages of the two channels the first and third tube should be the phase splitters while the centre ECC83 will be used for the two first stages. For 2 per cent. intermodulation distortion the amplifier has an output of 17 watts. Harmonic distortion is well under .5 per cent. at this point and both sides of the

sine wave begin to break up symmetrically at 17 watts. Frequency response is within 1 decibel from 17 c.p.s. to 85 kc/s with the output dropping 3 decibels at 12 and 64,000 c.p.s. Adjustment of feed back for an optimum square wave at 10 kc/s reduced the very high frequency output slightly with a smaller capacitor at C3 the half power point was raised to over 90 kc/s but in the interests of better square wave response and greater stability, the given value of C3 was chosen. For maximum output the input signal should have a minimum value of .1 volt which is completely satisfactory.

PARTS LIST

(Two each are required of all parts except those that are common to both channels. These are preceded in the list by asterisks).

- | | | | |
|-----|--------------------------------------|------|--|
| C1 | .05 microfarads 400 volts paper | R15 | 100 ohms 1 watt |
| C2 | 100 microfarads 3 volts electrolytic | *R16 | 75 ohms 10 watts |
| C3 | 470 peak farads 500 volts mica | *R17 | 300 ohms 20 watts |
| C4 | 0.1 microfarads 600 volts paper | *R18 | 100 ohms 5 watts wire wound |
| C5 | 0.1 microfarads 600 volts paper | *R19 | 100 ohms 5 watts wire wound |
| *C6 | ABC 40-40-40/450 electrolytic | *R20 | 27k 1 watt |
| *C7 | ABC 30-30/450,125/25 electrolytic | *R21 | 27k 1 watt |
| R1 | 470k $\frac{1}{2}$ watt | *R22 | 27k 1 watt |
| R2 | 470 ohms $\frac{1}{2}$ watt | T1 | Partridge P5201 Anode to Anode load
9,000 12,000 ohms
Secondary 4 equal 1 ohm windings |
| R3 | 22 ohms $\frac{1}{2}$ watt | *T2 | Power transformer
400-0-400 volts at 200 milliamps
6.3 volts centre tap at 5 amps
5 volts at 3 amps |
| R4 | 120 K ohms 1 watt | | |
| R5 | 1.2 Meg. $\frac{1}{2}$ watt | | |
| R6 | 68k 1 watt | | |
| R7 | 4.7k $\frac{1}{2}$ watt | | |
| R8 | 100k 1 watt | | |
| R9 | 100k 1 watt | | |
| R10 | 330k $\frac{1}{2}$ watt | | |
| R11 | 330k $\frac{1}{2}$ watt | | |
| R12 | 1k $\frac{1}{2}$ watt | | |
| R13 | 1k $\frac{1}{2}$ watt | | |
| R14 | 100 ohms 1 watt | | |

Valves

- 4 MULLARD EL34
- 2 MULLARD ECC83
- 1 MULLARD EZ34

CHAPTER V

STEREO PICK-UPS

We would like to thank Cosmocord, the makers of Acos pick-ups, for their kind permission to publish the following chapter which originally appeared in their news letters in a slightly different form.

Stereophonic reproduction can be looked on from artistic, commercial and technical viewpoints. From an artistic point of view, there is no doubt that stereo is a great improvement on monaural reproduction, even on playback equipment that has restricted frequency response compared with the best monaural reproducers the results seem more natural. For example, there is a spread of sound which, together with an increased realisation of the inner parts of the orchestra, makes the reproduction much more natural. Where the frequency range is comparable, any high-fidelity screaming strings with which we are all too familiar on some types of playback equipment (due to resonance in either the pick-up or speakers) are entirely absent on stereo. A stereo system can be operated at a lower level of sound intensity without losing the sense of reality, whereas with monaural systems there is always a sudden position in the volume level where the music as it were loses its presence and impact. Increasing the volume with stereo seems to move the listener nearer to the orchestra. It is a fact that if practising musicians are left to their own devices they will invariably play a high-fidelity reproducer with a considerable amount of top cut in order to achieve what, to their ears, is a realistic frequency response although this form of set-

ting to the high-fidelity addict is an anathema. Maybe artistic and technical ears will now meet in stereo.

From the technical viewpoint the stereo record carries two channels of information, that is to say, the single groove has two modes of movement. If the source of any particular sound is coming through from say the right, then the right hand microphone will receive a stronger signal than the left hand one, this is exactly how human ears work. The relative strength of sound received in each ear gives sense of direction and so with stereo recording you can get direction, depth and perspective. The two outputs from the two microphones are fed via two amplifiers to a recording head or cutter. Now the sound from the right hand microphone will cause the whole groove to move up and down in the direction shown in Fig. 7. The output from the right hand microphone will cause the groove to move up and down in the direction shown in Fig. 8. It should be understood that there will always be an output from both microphones unless the recording is being deliberately faked. At any particular instant the groove will be moving in a certain direction and this can be resolved as a motion partly as indicated in Fig. 7 and partly as indicated in Fig. 8. An exact parallel is given

by the motion of a car going Northeast with a town directly to the North and another to the East. The car will be travelling partly to the North and partly to the East. In other words the journey can be resolved in terms of Northwards and Eastwards directions. This is illustrated in Fig. 9.

In the Acos stereo cartridge, the stylus is connected by mechanical linkage to two crystals so arranged that one crystal will respond only to movements due to the left hand channel, and the other one will respond to movements of the right hand channel. The output from the two crystals are fed through two identical amplifiers which in turn are connected to two identical loudspeaker systems. We can now consider the requirements for the stereo cartridge, its presentation, connection and demonstration.

As with a single channel system, stereo reproduction can only be as good as the pick-up which must have a wide frequency response, be free of resonance and have low distortion. In addition, and this is where the stereo cartridge has extra requirements, there must be good separation between the two channels, the two outputs must be equal in all respects. There must also be good compliance both laterally and vertically. The separation between the two channels will not be the same throughout the frequency range, but it should be better than 15 decibels. Response curves show less than this above 8 kc/s but this is not necessarily due to the cartridge because test records have varying separation. Apart from the fact that resonances in the upper register will make the reproduction shrill, particularly above

The output from the right-hand microphone will cause the groove to move up and down in this direction.

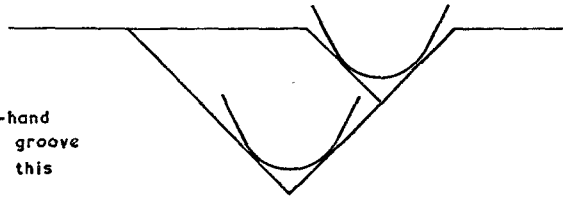


Fig. 7.

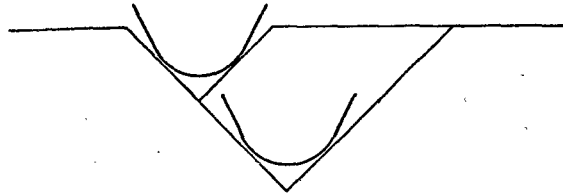


Fig. 8.

NORTH TOWN

START

EAST TOWN

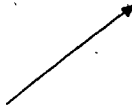


Fig. 9.

9 kc/s it will also have the effect of greatly diminishing the separation between two channels, just where the wide separation is particularly required. It is as well to remember that whilst the lower register of both the Acos stereo cartridges is remarkably smooth, serious resonances can be introduced by the design of the pick-up arm, a point we will deal with later. Good separation between the channels means that the output from one channel must contain very little of the recorded intelligence of the other channel, and vice versa, otherwise the stereo effect will be lost and at the worst there will be some sounds from both speakers. Intermodulation is the influence of one frequency upon another. An example from real life is the effect obtained when one talks above the tolling of a loud bell, or in a noisy aircraft, one's voice ceases to be heard clearly and has a sort of burbling sound upon it.

Pick-ups, amplifiers and loudspeakers can all introduce intermodulation distortion by themselves, but if there is a bad distortion in the pick-up it will be amplified by the rest of the chain. This form of distortion in pick-ups is extremely difficult to measure but can be shown if two types, one with bad intermodulation distortion and the other with very little are played side by side. The first model will give a muddled sort of tone and bad definition of the middle of the orchestra for the simple reason that frequencies are produced which are not harmonically related. In other words, you get unnatural sounds in the literal sense. The other pick-up will produce clean, treble with good definition. This explains why some high-fidelity pick-ups have a very wide frequency response but with intermodulation distortion sound so unpleasant compared with pick-ups having less wide range but with no intermodulation distortion. It is interesting to note that intermodulation distortion increases with tracking error measurements taken in the Acos Laboratory with both their stereo pick-ups show that intermodulation distortion to be less with that of comparable monaural types. This subject has been mentioned at some length because it has become quite a fashionable thing to talk about it and it is a form of distortion

which is the least understood.

Acos stereo cartridges, being crystal, are not prone to hum pick-up and special precautions need not be taken in this respect.

CONNECTION

It is true to say that one merely substitutes a stereo cartridge for a monaural one but there are one or two points over which care should be exercised. The Acos stereo 71 cartridge has three terminals, the centre one of which is earth, and the Acos stereo 73 cartridge has four, the two centre ones which are earth connections. Four terminals are being provided on the 73 at the request of manufacturers and also to prevent conversions. Normally one will short the two inner terminals together, in fact it is intended to supply a shorting ring. We strongly advise using a separate screening cable for each channel as there is a risk of cross talk being introduced if one uses a pick-up lead consisting of two inner conductors and one screened outer. It has been learnt that one automatic changer manufacturer is marketing his home models with the latter type of lead and the export model with the former type, and it will probably be as well to investigate which type of lead is used if a stereo cartridge is to fitted into the changer or player advertised as "wired for stereo". Ordinary rules then apply, that is, the pick-up lead should be kept as short as possible and the capacity of the lead should not be more than 100 pfs, which comes out in practice as using thin screened wire. Heavily screened cable is bad in any case because it will impede the lateral movement of the pick-up arm. It is most essential that the pick-up arm can move freely laterally and vertically. This has been stressed by many people from time to time but so often pick-up arms are not really free. Stiff pick-up arm movement has a very sad effect, especially on a stereo set up, most automatic changers and record players leave the manufacturers with reasonably free movement but long use in dust laden atmospheres, and general lack of maintenance, leave their mark. Check the level of the turntable with a spirit level. The actual fitting of a stereo cartridge into the pick-up

arm in the case of the 73 series, is by means of a bracket and merely becomes a matter of substitution for the existing single channel cartridge. Both stereo pick-ups have their elements so connected that phasing is correct for both stereo and single channel records, which is not the case with some American cartridges. In all cases it will be necessary to check the weight. You will probably be able to track on as low a pressure as 3 to 4 grams, it is not recommended to track above 8 grams. As a matter of interest, both the 71 and 73 cartridges, together with a precision made pick-up arm, will successfully track at 2 grams. Ordinary single channel records sound far better played

with a stereo pick-up through two channels, some manufacturers arrange their circuitry to play single channel records with an ordinary monaural pick-up both channels being in parallel. This we feel is a mistake, the Acos LP/Stereo stylus is equally suitable for both groove widths and nothing is to be gained by not using the stereo pick-up.

Decca and E.M.I. are producing stereo records of very high quality. They are in the shops and cost the same as ordinary L.P.'s in the United Kingdom, a remarkable achievement and both companies have shown great vision in not increasing the price of stereo discs over ordinary L.P.'s.

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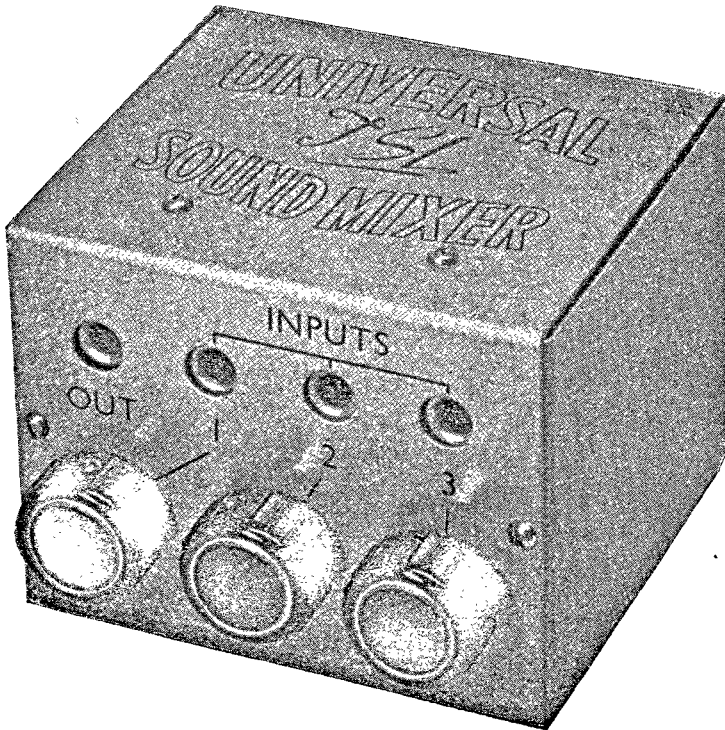
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