PRACTICAL STEREO AND QUADROPHONY HANDBOOK

BY B. B. BABANI

No. 208 BERNARDS (publishers) LTD 75p

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The Grampians
Shepherds Bush Road
London W6 7NF

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

I. S. B. N. 0 900162 43 0

First Published - February 1975
Reprinted - July 1976

Printed in Great Britain by C. Nicholls & Company Ltd

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TO HELP OUR EUROPEAN AND AMERICAN READERS WE INCLUDE HEREWITH A FULL EQUIVALENTS AND INTERCHANGEABILITY LIST OF THE SOLID STATE DEVICES USED IN THE CIRCUITS SHOWN IN THIS BOOK.

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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 ro 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 degrees to one side of the vertical and the other element responding to stylus motion at a 45 degrees 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 degrees to the vertical, so that the angle between the two sides is 90 degrees. 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 degree 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 degress to right of vertical and the other corresponding to stylus motion at an angle of 45 degrees 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.

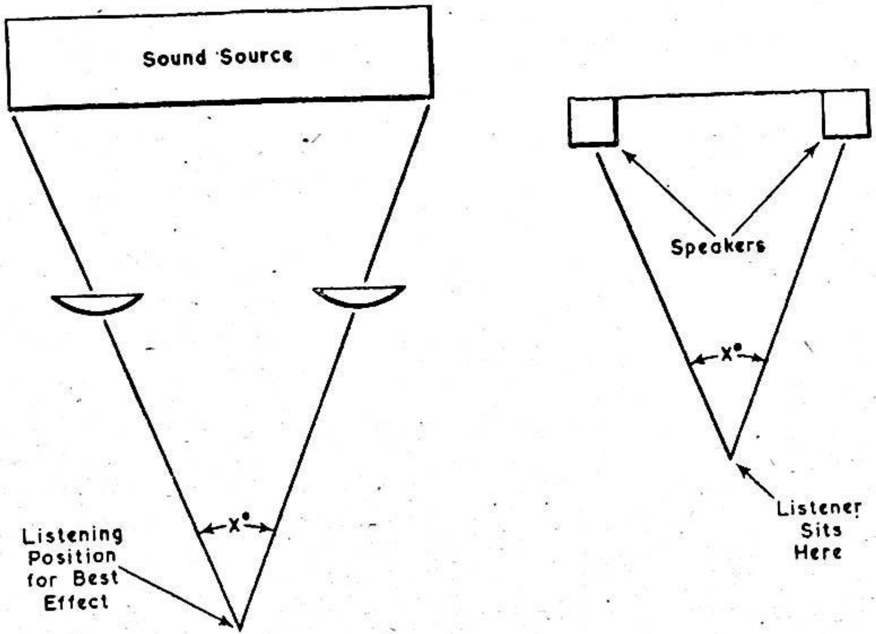


Fig. 1 Listening Angle Principle

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 times 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 degrees on one channel. This usually means merely interchanging the two leads to one of the speaker system. 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 parellel 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 some times 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.

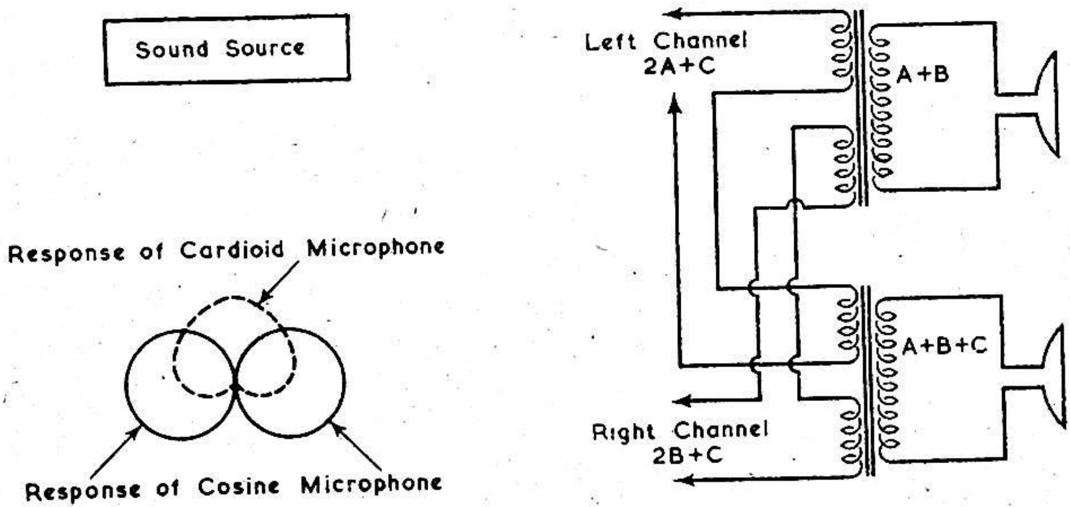


Fig. 2 Mid-Side Stereophonic Recording

4. Midsight 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 degrees out of phase with the (b) sound inas much as the microphone has but one pressure 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.

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 decibles 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 decibles down.

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 understnad 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

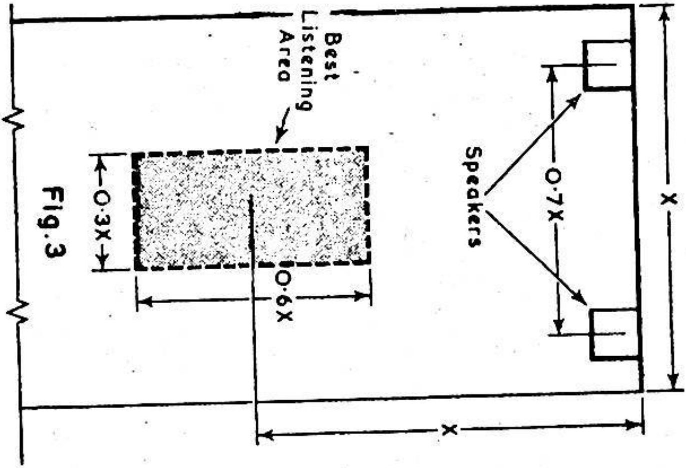

Best
Listening
Area
Speakers
X
0·7X
0·3X
0·6X
x

Fig. 3

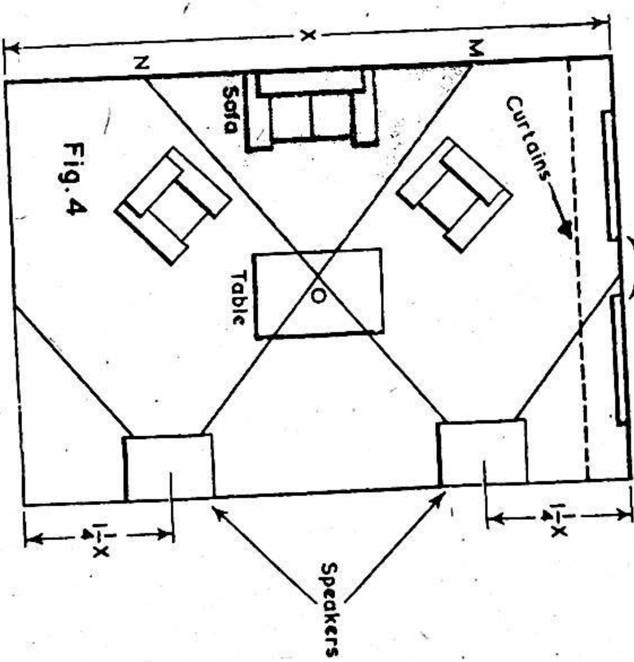
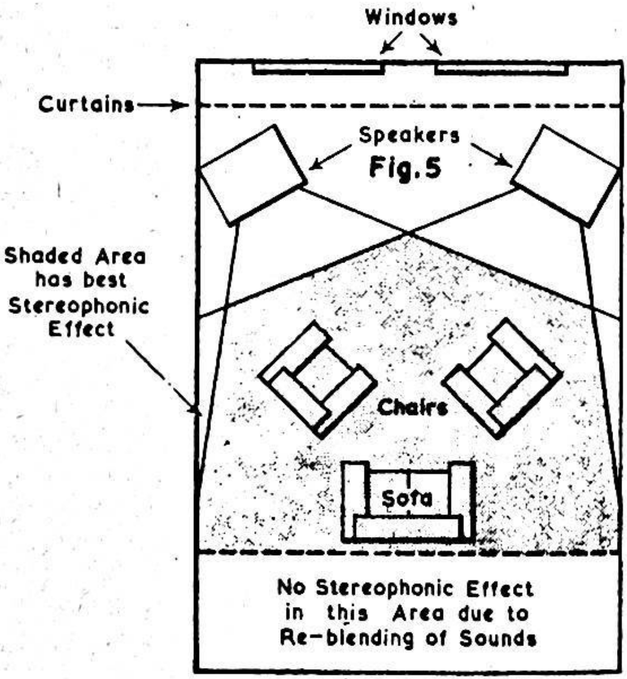

X
M
N
Sofa
Curtains
Windows
Table
O
$\frac{1}{4}$X
$\frac{1}{4}$X
Speakers

Fig. 4

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 posisitons.
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, courches, 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 $\frac{2}{3}$rds x $\frac{1}{3}$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 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 M N O now receives an admirable stereophonic effect and other points outside this area, separate 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 degrees 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 theatré 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 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 sette is, moving the settee forward might improve the results in certain types of room. The existence of doors might make this arrangement

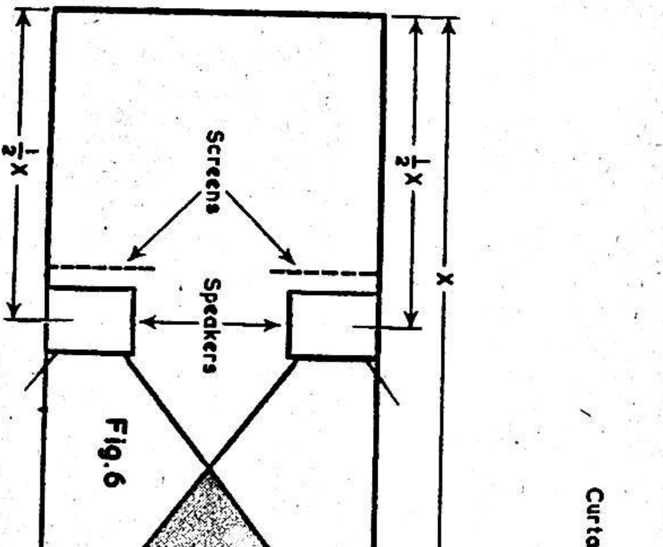

Screens
Speakers
$\frac{1}{2}$X
$\frac{1}{2}$X
X
Curtains
Chairs
Table
Window

Fig.6

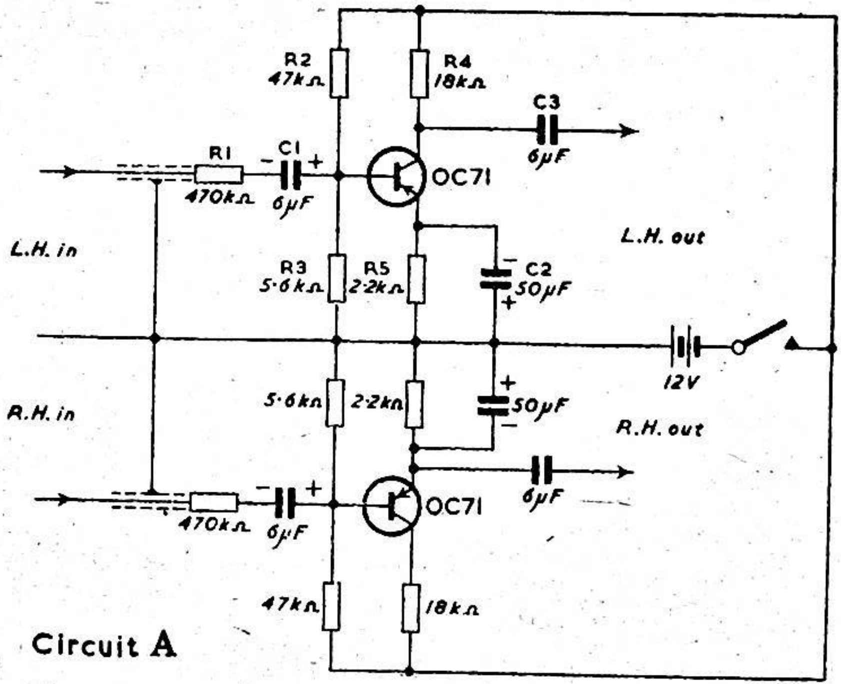
impossible, however, a slight re-arrangement is not likely to upset the stereophonic effect too much.
In certain cases it may be difficult or impossible to sit exactly halfway between the two speakers. This is not absolutely essential, what is essential is that the levels of volume arising from the two 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.

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 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 which are available specially wired for stereo.
As the output required by the headphones is so low a transistor amplifier offers several advantages and several varieties of this follows.
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 A

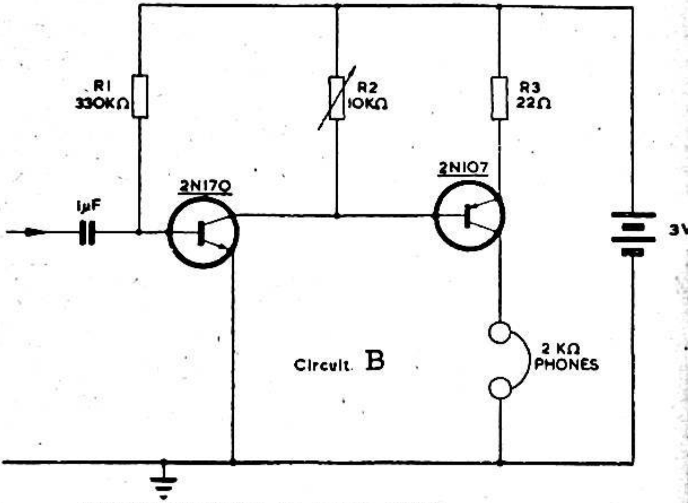
The circuit shown here uses transistors with the following advantages.
As no heaters are used there is no possibility of hum being introduced and with headphones the slightest hum is immediately noticeable.
The transistors shown are OC71 but almost any type of small signal PNP transistor would be equally 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


R1
470kΩ
C1
6μF
R2
47kΩ
R4
18kΩ
C3
6μF
OC71
L.H. in
L.H. out
R3
5·6kΩ
R5
2·2kΩ
C2
50μF
12V
5·6kΩ
2·2kΩ
50μF
R.H. in
R.H. out
6μF
470kΩ
6μF
OC71
47kΩ
18kΩ

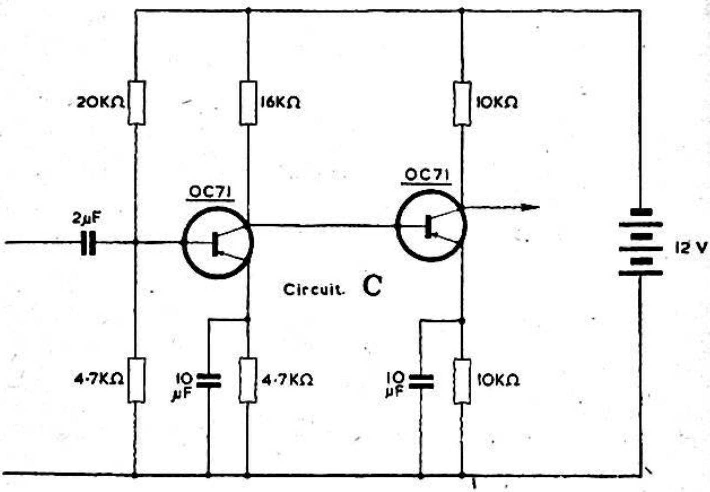
Circuit A

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 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 impredance 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 it 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

As was mentioned when discussing Circuit A occasionally a single transistor amplifier provides insufficient gain to drive the headphones. Circuit B 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. 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



R1 should be adjusted for optimum results



Direct–Coupled A.C. Amplifier

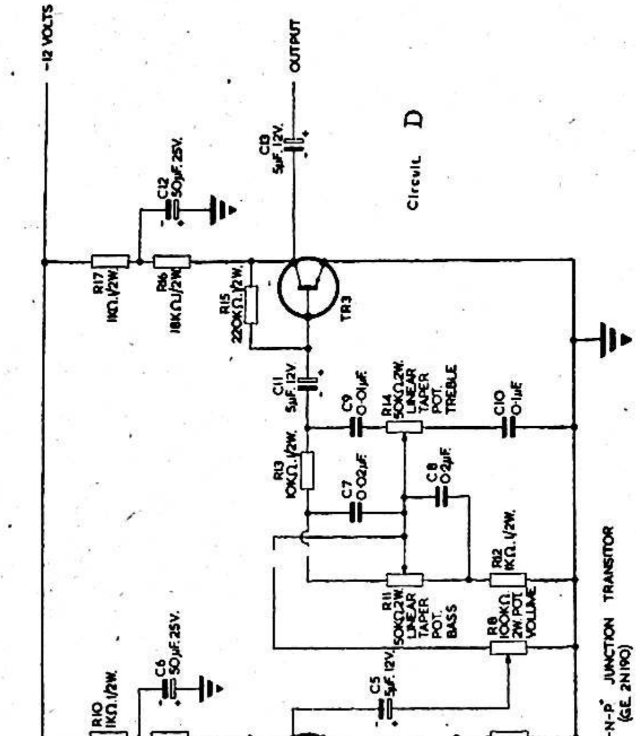
collector current for TR1 is partially supplied via TR2. When using two such amplifiers for stereo, the two 10 K ohm resistances should be ganged together so that the volume control operates both amplifiers at the same time. R3 which, as may be seen from the diagram, is in the emitter lead of TR2, is 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.

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 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 D

Circuit D 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 res ponse 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


-12 VOLTS
OUTPUT
C13 5µF. 12V.
C12 50µF. 25V.
Circuit D
R17 1KΩ. 1/2W.
R16 18KΩ. 1/2W.
R15 220KΩ. 1/2W.
TR3
C11 5µF. 12V.
C9 0·01µF
R14 50KΩ 2W. LINEAR TAPER POT. TREBLE
C10 0·1µF
R13 10KΩ. 1/2W.
C7 0·02µF
C8 0·2µF
R12 1KΩ. 1/2W.
R11 50KΩ 2W. LINEAR TAPER POT. BASS
R8 100KΩ. 2W. POT VOLUME
C5 5µF. 12V.
C6 50µF. 25V.
R10 1KΩ. 1/2W.
R6 18KΩ. 1/2W.
R5 220KΩ. 1/2W.
TR2
C4 0·05µF
R7 220Ω. 1/2W.
R9 1KΩ. 1/2W.
R3 18KΩ. 1/2W.
C2 5µF. 12V.
R4 4·7KΩ.
R2 220KΩ. 1/2W.
TR1
C3 50µF. 25V.
C1 5µF. 12V.
R1 3·9K. 1/2W.
PICK-UP
TR1, TR2, TR3. - "P-N-P" JUNCTION TRANSITOR (GE 2N190)

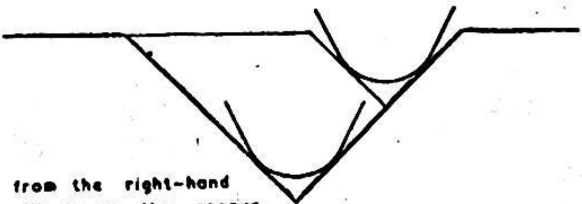
and applied to a resistor in the cathode of the first stage. This method of feed back is not well suited for an all-transistor amplifier since voltage feed back tends to control the voltage applied to the next stage whereas it would be more desirable in transistor amplifiers to control the current in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

If a transistor pre-amplifier is to be used with a valve amplifier however, voltage feedback 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 are 2N190 a suitable replacement would be the OC70 for TR1 and TR2, and the OC71 for TR3.

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 familar 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 low level of sound intensity without losing the sense of reality, whereas with nonaural 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 setting 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 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.



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

Fig 7.

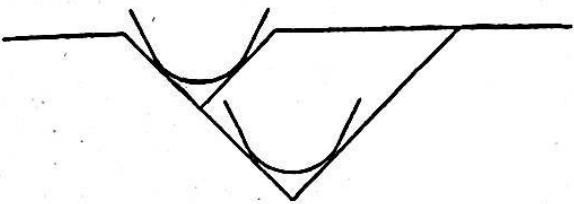


Fig 8

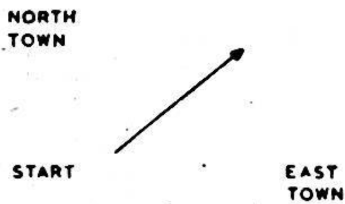


Fig 9

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 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 clealy and has a sort of burbling sound upon it.

Pick-ups, amplifiers and loudspeakers can all introduce intermodulation distorition 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 distorition 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 converstions. 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 fit 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 practise 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 pickups 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 as pressure as 3 to 4 grams, it is not recommended to track about 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.

THE TEN-PLUS-TEN STEREO AMPLIFIER

SPECIFICATIONS:

A transistorised stereo amplifier capable of being fully driven by ceramic cartridges or other "flat" signal source with an output of at least 100mV. Output and regulator transistors are protected against damage.

Power: 10 watts music power or 8 watts RMS per channel into 8 ohms, each channel driven separately or both together; 5 watts RMS per channel into 16-ohm loads, 4-ohm speakers may not be used.

Distortion: Less than 0.85 per cent at 1KHz at 8 watts; less than 0.5 per cent at 1 watt at 1KHz.

Signal-to-noise ratio: 58dB with respect to 8 watts.

Separation between channels: −43dB or better, with respect to 8 watts at any frequency between 100Hz and 10KHz.

Frequency response at 1 watt:– 3dB points at 25Hz and 17KHz (−4dB at 20KHz).

Tone Control: 13dB cut at 10KHz.

Input impedance: 1 megohm, all inputs.

Many systems have been suggested for protecting the output stages in an amplifier. For the most part, however, the ideas were not very appropriate for use with an economy stereo amplifier. Since then we have developed a protection circuit which is incorporated into the power cupply. It is effective, easily reset after overload—and economical.

In operation, if the amplifier is overloaded by a short-circuit or is grossly overdriven, the power supply to the amplifier is automatically disabled. To restore the amplifier to working condition, the cause of the overload is simply removed, the amplifier switched off for ten seconds or so, and then switched on again. Circuit operation will revert to normal.

Four transistors are used in each power amplifier, one silicon type for initial amplification, and three germanium types as the driver stage and output pair. The input transistor is a silicon NPN type which provides voltage amplification. Negative AC and DC feedback is applied to the emitter from the emitter resistors of the output transistor pair.

The quiescent voltage at the junction of the output transistor emitter resistors should be about half a volt less than half the supply voltage, which it will normally be with standard tolerance components. If need be, it can be set, with the aid of a suitable voltmeter, by varying the 1.5K emitter resistor of the first transistor. Increasing the resistor will increase the voltage and vice versa. When set thus, a sinewave signal should clip symmetrically at the point of overload, thus resulting in maximum power before the onset of clipping. If the necessary equipment is available, operation can be double-checked by observation of the waveform with an oscilloscope.

The class-A driver stage is a germanium PNP transistor, AC128, operating with a collector current of 50-mA. At this current the usual flag heatsink is insufficient and the flag heatsink must be mounted on an aluminium plate which, inturn, is attached to the printed board.

The output transistors are a complementary germanium pair AD161/162 operated in the normal push-pull complementary-symmetry mode. These transistors have the advantage of being able to deliver high power with a relatively low supply voltage, because of low IR losses in the transistors themselves. The quiescent current of the output stage is nominally 10mA which, again, will not normally require setting up. If it does prove to be necessary, however, it can be altered by variation of the 68-ohm resistor between the two output transistor bases. Increasing the resistance will increase the current and vice versa.

Note that the quiescent current of the output stage cannot be set unless a load is connected, since the load forms part of the biasing network for the output transistors.

However, the sensitivity for full output is of the order of 150mV, with an input impedance of 60K. This means that an extra stage of amplification is required to obtain the necessary sensitivity and high input impedance to enable the amplifier to be driven to full output by the lower output, higher quality ceramic cartridges which constructors may well wish to use.

The input stage employs two transistors connected as a "Darlington pair". In this mode of operation, the emitter of the first transistor is connected to the base of the second and they share a common collector load. This provides high input impedance, the required gain and a low noise content.

The volume control follows the input stage, coupled to it via a 10uF electrolytic capacitor. The volume control will thus attenuate any noise generated in the input stage at its normal settings, making for lower background noise in typical use.

However, having the volume control in this position means that the input stage may be overloaded if fed with too large a signal. The stage overloads with an input signal of 2 volts RMS (sine wave) and this means it is not directly suitable for high-output crystal cartridges —as opposed to ceramic cartridges which usually have considerably lower output.

If a crystal cartridge must be used it should be connected via a high impedance divider, say 2 megohms and 270K. Alternatively, one can shunt the cartridge with a suitable capacitor (say 0.0047uF) which will reduce the output voltage, while improving the effective bass response. To obtain the best reproduction from the amplifier, however, one of the better quality ceramic cartridges should be used.

The power supply is basically a conventional full-wave rectifier system followed by a series regulator. When both channels of the amplifier are driven to full power, the total current drain is over 1.2 amps, while the current under "no-signal" conditions is of the order of 130 milliamps. This large variation in current drain, plus the fact that the output transistors have a relatively low collector-emitter voltage rating of 32 volts, means that a regulated power supply is mandatory.

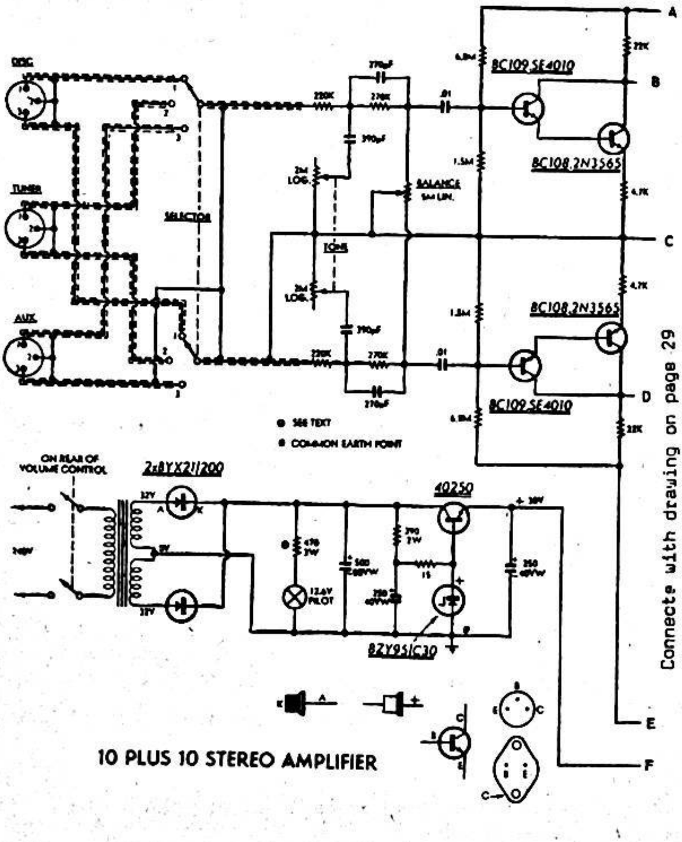
The transformer rectifier combination supplies 40 volts DC to the input of the regulator. Having two separate secondary windings, the transformer can be connected to suite a bridge rectifier or the conventional "centre-tapped winding" rectifier. Instructions on the method of connection are normally packaged with the transformer. We used two BYX21/200 20-amp automotive diodes as rectifiers these being about the cheapest high current diodes available. Off-setting their economy is the disadvantage that it is necessary to make a soldered connection to the case, when tends to be a little untidy.

The series regulator consists of a silicon NPN power transistor connected in the "emitter-follower" configuration. This takes care of voltage regulation, filtering and overload protection.

A low value resistor in the negative supply line senses the amount of current drawn from the supply. If the current exceeds a certain value, the voltage developed across the sensing resistor triggers the thyristor (in parallel with the zener diode) into conduction, thereby removing the forward bias from the transistor and interrupting the current drain. The actual "switch-off" time is dependent upon the particular transistor and thyristor but can be expected to be of the order of a couple of micro-seconds. This should provide adequate protection against even the most catastrophic overload.

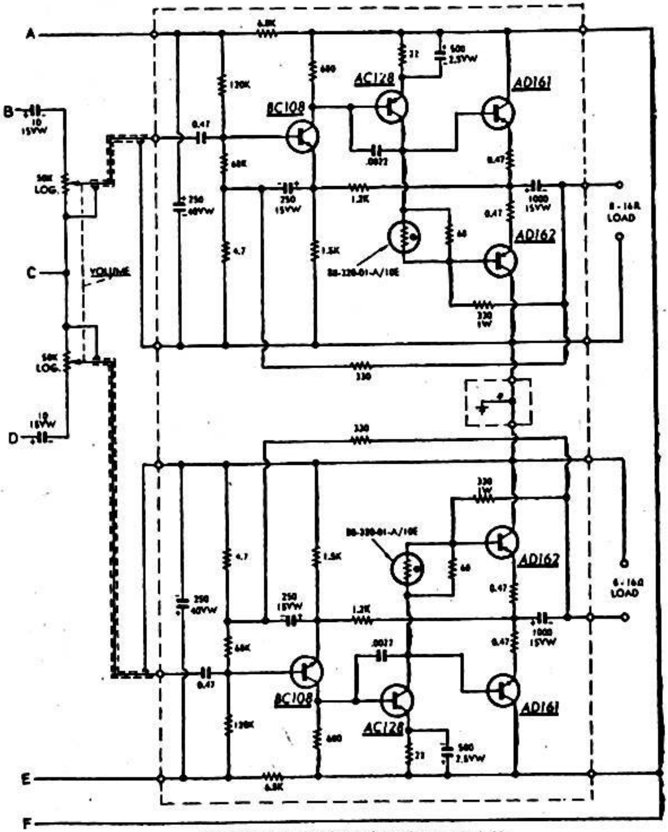
The value of current at which the thyristor is triggered should lie between the maximum peak current drain of about 1.7 amps and the maximum current rating of the output transistors, which is 3 amps. The optimum current at which the thyristor triggers might thus be nominated as 2 amps. The maximum triggering voltage of the C106Y1 thyristor used is 0.8 volts so that a sensing resistor of 0.39 ohm would seem to be a suitable choice. However, individual C106Y1 thyristors may trigger, in this circuit, at voltages as low as 0.45 volts which would mean premature triggering if a sensing resistor of 0.39 ohms is used. For this reason, we have specified 0.39 ohms but with the proviso that it will have to be shunted down, in most cases, so that the thyristor triggers at about 2 amps.

The value of the shunting resistor can be determined with an ammeter and a load resistor, connected across the supply to draw 2 amps. Alternatively, it may be selected so that the thyristor triggers when both channels are driven to just over full power, i.e., past the onset of clipping. If facilities are not available to measure power output or observe clipping, the shunting resistor can be selected so that the thyristor triggers when the amplifier is driven to loud levels on normal signal.



10 PLUS 10 STEREO AMPLIFIER

Connects with drawing on page 28



CIRCUITRY WITHIN BROKEN LINES IS ON PRINTED WIRING BOARD

10 PLUS 10 STEREO AMPLIFIER
(with overload protection)

The power diode connected between the zener diode and the base of the transistor ensures that the transistor is turned completely "off" The thyristor remains in a state of conduction after the transistor is turned off, supplied with current via the 390-ohm and 15-ohm resistors. The only way that normal operation can be restored is to stop the current through the thyristor by switching off the mains supply for about ten seconds, which allows the filter capacitors to discharge. Normal operation can then be restored by switching on again.

The ripple content from the supply, even at maximum current drain, is quite low due to the capacitance amplification of the regulator. However, while the effective capacitance of the supply is very high in terms of filtering, the intrinsic output impedance of the supply is not as low as might be desired, due to the sensing resistor and the dynamic resistance of the collector-emitter junction of the transistor. To reduce the effective output impedance of the supply, a 250uF capacitor has been specified across the output.

It may be thought that the 250uF capacitor could cause damage to the regulator transistor at "switch-on", due to the fact that it would act as a temporary short-circuit and is not included in the loop which is monitored by the sensing resistor. In fact, the time-constant formed by the 390-ohm bias resistor and the 500uF capacitor in the base circuit of the transistor causes it to turn "on" gradually, effectively limiting the surge current. For the same reason, there is no "plop" from the loudspeakers as the amplifier is turned on–an effect which is noticeable with some transistor amplifiers.

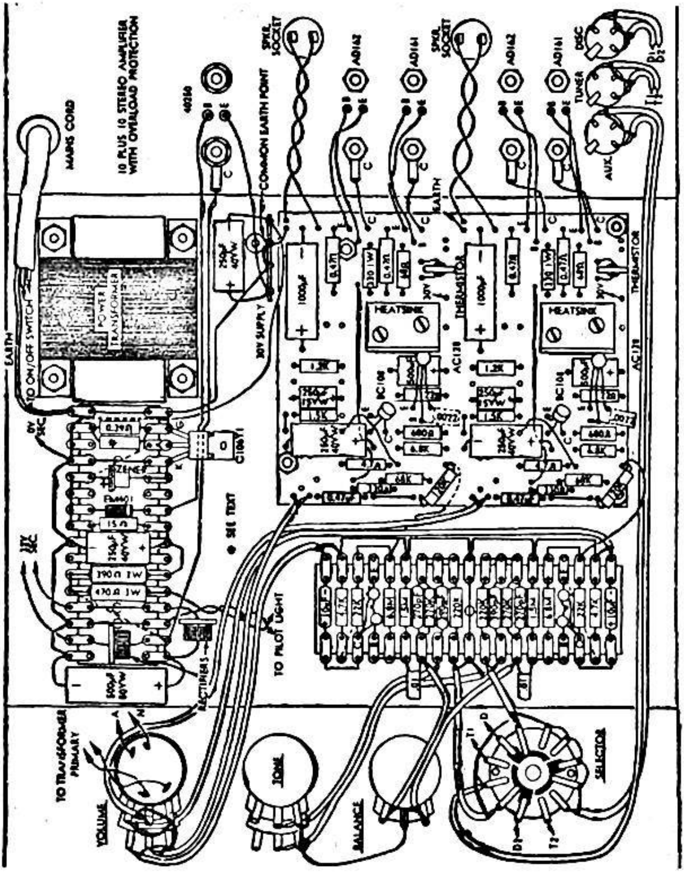
CONSTRUCTION: The amplifier is assembled in a chassis with overall dimensions of 9¾ x 7-7/8 x 3¼ inches. It is U-shaped, with a ½-inch flange all round. The prototype was made of 18-gauge aluminium. We would not advise a reduction in the metal thickness, as it would reduce the potential effectiveness of the chassis as a heat-sink for the power transistors.

The four output transistors are mounted on the rear of the chassis, mica washers being used to insulate them electrically from the chassis. If the chassis has been painted, the area to which the transistors are mounted must be rubged back to bare metal to ensure efficient heat transfer.

Actually, under normal "programme" conditions, "heatsinking" requirements are not exacting and, even on hot days, the output transistors will be merely warm to the touch. However; we would not advise extended full power testing on hot days, since the chassis– and of course the output transistors–will become rather warm.

If an application is envisaged where the full RMS power of the amplifier is used continuously, a more efficient heatsink for the transistors should really be provided. For normal music reproduction in the home, the chassis itself will be quite adequate. however.

When purchasing the transistors, be sure to obtain the mica washers and nylon bushes, the latter being required to insulate the retaining



The above wiring diagram and the accompanying photographs will make construction straightforward. Do not forget the mains cord clamp which is not shown above. The amplifier earth returns should be connected to the chassis at the common earth point.

nuts from the chassis. When mounting the transistors, silicone grease should be lightly applied to both the transistors and the heatsink (chassis). Connection to the collectors of the transistors is made to a solder lug placed under one of the retaining nuts. The same mounting instructions apply for the regulator transistor.

Polarised two-pin sockets are used for loudspeaker connection, since they simplify speaker phasing and are easier and safer to use than terminal lugs. Three 3-pin DIN sockets are provided for the inputs and these are connected to the selector switch via dual shielded cable. The shield for each channel input is connected to the centre pin (pin 2) of the DIN socket but no connection is made to the outer shield of the socket.

All the circuitry for the input stage is mounted on a piece of miniature tagboard.

Similarly, the power supply circuitry, apart from the transistor and 250uF capacitor at the output are mounted on tagboard. The thyristor is soldered directly to the tagboard, no heatsink being necessary. The electrostatic screen of the power transformer and the earth wire of the mains cord are connected directly to the single common earth point on the chassis, via the tagboard. They should not be connected to chassis via the current sensing resistor. The 250uF capacitor is mounted on a four-way tagstrip which also serves as the common earth point to the chassis. The negative connection for this 250uF should not be made directly to earth but via the sensing resistors.

The pilot lamp is supplied from the output of the rectifiers via a 470-ohm 3-watt resistor. This will have to be varied according to the voltage rating and current drain of the lamp. The lamp in the prototype drew about 70 milliamps at 8 volts–it was rated at 12 volts. It is good practice to run these lamps at reduced voltage to obtain longer life and reduce brightness so that it is not obtrusive. A 560-ohm resistor should be suitable for typical 6-volt lamps.

The third pole of the 3-pole selector switch is used for shield terminations. As the circuit and wiring diagrams indicate, only one of the two shields of the dual input cables and the signal cable to the input stage is connected to the third pole of the selector switch, to decrease the possibility of earth loops.

The input wiring sheild must not be connected to the chassis at any point apart from the connection made via the printed boards.

All the above wiring details are shown on the wiring diagram. It is good practice to check your wiring against the circuit diagram. If the details of the wiring layout are not noted and duplicated, instability or, at the very least, failure to obtain the performance of the prototype, may result.

The power transformer used in the prototype was rated 32 volts at 2 amps DC.

A suitable order of assembly would be as follows: First, wire all the components and connecting wires into the printed board. The 2200 pF capacitor connected from base to collector of the driver transistor is wired on the copper side of the board, as is the positive supply connection for the input stages. The heat sinks for the driver transistors are made of 18 gauge metal, 1¼ inches wide by 2-3/16 bent at right-angles 13/16 inches from the end, with holes drilled for mounting screws. The two tagboards are also pre-wired as noted previously.

Note that all the boards are mounted on the chassis so that they have at least ¼-inch clearance of the same. This can be done using fibre rod spacers tapped right through for 1/8-inch whitworth or, alternatively, using just 1/8-inch screws and nuts.

Having wired the boards, attention must be given to the chassis. The rubber feet are retained with a screw and nut, the nut being held in the foot itself. The potentiometer and selector switch shafts should be cut to suit the knobs. Having installed the controls and input sockets, transformer and rectifier, etc., the appropriate wiring can be installed. The mains cord should be passed through a grommeted hole in the rear of the chassis, then between the chassis side flange and the transformer stack and finally terminated at the switchpot. It is anchored by a clamp held by the same screw which retains one of the transformer lugs.

The five power transistors may now be mounted, as detailed earlier. Then install the supply board and make inter-connections to it. Next, install the printed wiring board and 4-terminal tagstrip for the earth and supply connections.

The quiescent current is most easily checked at this stage. It should be between 65 and 75 milliamps for each channel. Next, the input stage tagboard can be installed. The "half-supply" voltage at the junction of the emitter resistors of the output transistors may now be checked and adjusted if need be, as described earlier. The final adjustment is that of the sensing resistor which is described above.

Finally, a front panel and knobs can be fitted. The prototype had a panel with a glossy black background and white lettering which contrasts with the turned aluminium knobs.

The amplifier may be installed in a cabinet or used in a free-standing situation, in which case a cover is required. In both cases care must be taken to ensure that air can circulate freely around the rear of the chassis. The cover was made of 18-gauge aluminium, suitably bent and with a slight overhang at front and rear. Ventilation in the form of a row of holes at top and sides or louvres, must be provided, to allow the interior of the amplifier to remain cool. The transformer was the main source of heat in the prototype, and if this is not allowed to escape the case can become quite warm to the touch.

Used with a high quality ceramic cartridge and good quality speakers the amplifier is capable of really excellent sound.

PARTS LIST

1 chassis with overall dimensions 9¾ x 7-7/8 x 3¼ inches made of 18-gauge metal
1 metal cover with dimensions to suit chassis (optional)
1 front panel
1 power transformer, 240V to 32V at 2A DC
3 3-pin DIN sockets
2 2-pin polarised sockets
Miniature tagboard, 1 19-lug, 1 16-lug
1 3-pole 3-position rotary switch

SEMICONDUCTORS
2 AD161/162 complementary matched transistor pairs
1 40250 silicon NPN transistor
2 BC109, SE4010 or similar low-noise silicon NPN transistor
4 BC108, 2N3565 or similar silicon NPN transistor
2 AC128 germanium PNP transistor (with heatsinks)
2 BYX21/200 power diodes or MB1 bridge rectifier

1 BZY95-/C30 zener diode
1 C106Y1 thyristor
2 B8-320-01A/10E or E215AB/15E thermistors

POTENTIOMETERS
1 50K (log dual ganged, with rotary power switch
1 2M (log) dual ganged
1 5M (lin)

RESISTORS (½ or 4-watt 5% tolerance)
2 x 6.8M, 2 x 1.5M, 2 x 270K, 2 x 220K, 2 x 120K, 2 x 68K, 2 x 22K, 2 x 6.8K, 2 x 4.7K, 2 x 1.5K, 2 x 1.2K, 2 x 680 ohm, 2 x 68 ohm, 2 x 22 ohm, 1 x 15 ohm, 2 x 4.7 ohm
(½-watt unless specified)
1 x 470 ohm/3 watt (to suit pilot lamp)
1 x 390 ohm/2 watt
4 x 330 ohm (2 x 1 watt, 2 x ½-watt)
2 x 0.47 ohm, 1 x 0.39 ohm

ELECTROLYTIC CAPACITORS
2 x 1000uF/15VW, 1 x 500uF/50VW, 4 x 250uF/40VW, 2 x 500uF/2.5VW, 2 x 250uF/15VW, 2 x 10uF/15VW

CAPACITORS
(Low voltage polyester, polystyrene or metallised polyester)
2 x 0.47uF, 2 x 0.01uF, 2 x 2200pF, 2 x 390pF, 2 x 270pF

SUNDRIES
4 turned aluminium knobs, 4 rubber feet, 8 x ½in fibre rod spacers, tapped right through for 1/8-in Whit. screws, 1 4-terminal tagstrip, mains cord and plug, mains cord clamp, grommet, miniature bezel and lamp, dual shielded cable, hook-up wire, spaghetti sleeving, screws, nuts, silicone grease, solder, etc.

NOTES ON THE 10-PLUS-10 STEREO AMPLIFIER

Several friends who have constructed the 10-plus-10 Stereo Amplifier, have written to me requesting information on various matters. Some of the points they raised are reproduced here in question and answer form, for the benefit of other constructors and readers in general.

Protection circuit: The first three questions involve the setting of the current sensing resistor for the thyristor overload protection circuit. The thyristor should be set to trigger at 2 amps.

(1) I found that the supply voltage drops markedly at current drains in excess of 1.2 amps. Is this normal?

This is quite normal. The regulator network is not intended to "regulate" at currents in excess of 1.2 amps, which is the maximum current drain under normal operating conditions. The main reason for the reduction of the supply voltage at high currents is that the base current demand of the power transistor rises to the point where it robs the zener diode of biasing current, so that the zener ceases to be a fixed reference voltage source.

(2) I used a dummy load across the supply to set the current sensing resistor. At a current drain of 2 amps the regulator transistor became very hot and it eventually failed after about 10 minutes at this current. A replacement transistor also became just as hot and I switched off the supply. I am wondering what is wrong?

When the amplifier is reproducing ordinary program material the current drain is small and the regulator transistor dissipates relatively little power – just a few watts. Rarely would the amplifier be driven to full power where the current drain rises to 1.2 amps. If the amplifier was overdriven to the point where the current rose to 2 amps the thyristor would be triggered and the regulator switched off. The point here is that, under normal conditions, the regulator dissipates little power and the chassis is an adequate heatsink. However, at a current drain of 2 amps, the transistor is dissipating in excess of 30 watts of heat and if this is maintained for more than a few minutes the transistor will be destroyed. If the regulator was intended for continuous operation at this power level we would have specified a higher power transistor and a more efficient heat-sink. In addition, the power transformer is only designed for a maximum continuous secondary current of 2 amps AC which corresponds to a DC current drain of approximately 1.4 amps. Thus, the power transformer is not intended to operate continuously at 2 amps DC from the regulator.

Why, then, did we originally select 2 amps as the current at which the overload protection thyristor was triggered? Because 2 amps represents a safe margin above the normal maximum current drain at which the five power transistors can operate safely – for very short periods – and above which they would easily be destroyed.

Frankly, we did not seriously consider the possibility that anyone would deliberately sustain the overload condition for an unbroken ten minutes.

(3) I have found that to be able to turn the volume control fully clockwise (when playing a typical record) without the thyristor triggering I have to use a very low value of sensing resistor. At maximum volume setting the supply voltage averages around 23 volts. It appears that the amplifier is drawing far too much current but nothing appears to be amiss. I am using dummy loads in place of speakers. Is the amplifier operating safely?

The fact that you are using dummy loads and that the voltage drops to 23 volts indicates that you are severely overdriving the amplifier. If you do use dummy loads, you should use a sine wave signal wource and monitor the output voltage with an AC voltmeter or oscilloscope. The thyristor can then be set so that the output voltage from both channels just exceeds 9 volts RMS as indicated by the voltmeter or when the oscilloscope shows the sine wave signal to be clipped. This will correspond to maximum power.

Alternatively, the current sensing should be set using an ammeter to monitor the current drain from the power wupply or using the listening test described in the article.

Quiescent Current: Another area of confusion concerns the measurement of the qyiescent current. The total "no-signal" or "quiescent" current of each channel should be between 65 and 75mA, while the quiescent current drawn by the output transistors is a nominal 10 mA. Some readers are confused as to the method of measurement of these currents, as the following question illustrate:

(4) The quiescent current of my amplifier (in series with the 80ohm loudspeaker) was about 35mA instead of 10mA as you quote. I also found that varying the 68-ohm resistor does little to alter the current which is contrary to your statement in the article. How is the current set to the correct value?

This question raises several interesting points, the first being that a small DC current flows through the loudspeaker and the second is that the amplifier will not function at all unless a load is connected. Apart from this, the current flowing through the loudspeaker is not the quiescent current drawn by the output transistors but a portion of the current flowing through the AC128 driver transistor. In fact, the loudspeaker forms part of the biasing network for the output transistors which is also the collector load for the driver transistor. Varying the 68-ohm resistor will cause a very small change in the driver transistor's load so that the measured current changes very little, as observed.

If the loudspeaker is removed, the driver transistor conducts less current and the output transistor current drops to almost zero. This means that, if high impedance headphones are used with this amplifier, a resistor with a value of 15 to 22 ohms should be connected across the output in place of the loudspeaker, in order that the amplifier should function normally.

The correct method of measuring the current though the output transistor is to measure the voltage across the emitter resistors and calculate the current using Ohm's law. Alternatively, an ammeter could be connected in series with the collector of the AD161. The current should be around 10mA as noted above. The current will be reduced to almost zero when the 68-ohm resistor is shorted out.

ERROR IN TEXT: In the text it states that the quiescent output voltage can be set "by varying the 1.5K emitter resistor of the first transistor". The sentence which follows is in error and should read: "Increasing the resistor will decrease the voltage and vice versa".

A CRITICAL LOOK AT QUADRASONIC REPRODUCTION

Quadrasonics – quadrasonics – quadrasonics! More and more space is being devoted to the subject in audio literature. It is a passing fad or does it herald a whole new era in high fidelity sound reproduc reproduction?

As the word more or less implies, "quadrasonic" refers to four-channel sound, as distinct from two-channel sound, which most hi-fi enthusiasts now enjoy and which generally goes under the name "stereo".

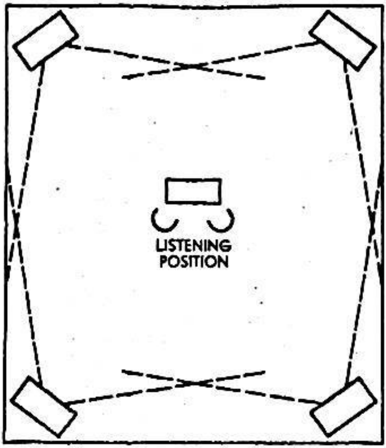
The idea of using more than two channels for sound reproduction in the home has gradually gained attention over the past couple of years. It has been called "four-channel stereo", "surround stereo" and other such names but the term which seems to be gaining favour in most quarters is "quadrasonic" (spelt with an "a" in the middle!)

One of the curious things about quadrasonics is that the growth of interest in the subject has been quite spontaneous – up to the present, at any rate. It emergence has not been the clear outcome of any commercial push or of any urgent, unfilled need on the part of hi-fi enthusiasts. There has been no sudden breakthrough in technology and no crusading by the journalistic fraternity. Rather does it seem that a mixture of all these elements, like a mixture of certain chemicals, has produced the end reaction – practical quadrasonic sound reproduction in the home.

The basic idea of quadrasonic sound reproduction has already been explained. Briefly, it assumes the recording of four spatially different versions of the original sound. These separate versions are reproduced through four separate amplifiers and four separate loudspeakers placed, typically, in the respective corners of the living-room.

The two loudspeakers fronting the listening position reproduce the direct sound which would normally reach the listener from the performer(s). The loudspeakers at the rear of the listening area are supposed to reproduce the echoes and the general "ambience" of the chamber in which the sound was recorded. The idea is to enhance the impression of being present at the original performance. This, at least, is the "purist" concept of quadrasonic reproduction.

During the past few years, tape technology has advanced to the point where it is now no longer a problem to impress and retrace eight or more tracks, side by side, on quarter-inch (or even narrower) magnetic tape. This is done, as a matter of course, on 8-track stereo cartridges.



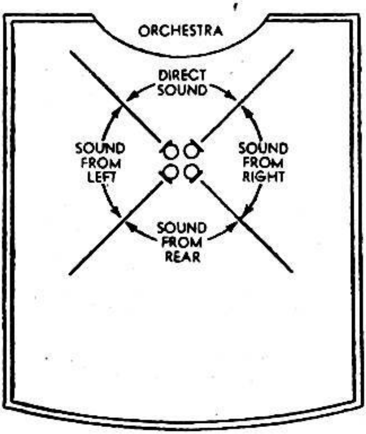
Quadrasonic reproduction in the home assumes the use of four distinct sound channels, with the loudspeakers located near the corners of the listening room. The system can be used to re-create concent-hall ambience or, in gimmick fashion, to project individual sounds towards the listener from any direction.

It has become practical to manufacture complex tape heads with four side-by-side gaps and four separate magnetic circuits, capable of tracing narrow parallel magnetic tracks; this without troublesome cross-talk and with an acceptable signal-to-noise ratio.

Tape technology having reached this stage, it was only natural that engineers and others should speculate about the possible application of four-channel rather than two-channel sound to domestic listening. It was equally natural that they should follow up such speculation with exploratory recording and listening sessions.

Since the job of technical writers is to report to their readers what is going on in their chosen field, news of these experiments soon got around. Readers were reminded of some of the forward looking sound effects employed in the Disney/Stokowski film "Fantasia", and others which followed it. It was an intriguing thought that a similar facility might soon be available in their own homes.

There was, of course, the expected wail: no sooner had hi-fi enthusiasts equipped themselves with expensive two-channel stereo equipment than this new thing had shown up to render it obsolete. The wails continue.



The "purist" concept of quadrasonic recording would presumably replace the listener in an optimum seat with four directional microphones, which would seek to recapture the all-round listening environment from that particular point.

But with the lament there has been a fair helping of crocodile tears. While one hand was extended in a gesture of despair, the other was already fumbling for a cheque book for the wherewithal to try out this new, more advanced and intriguing technique.

And when hands start fumbling for cheque books, commercial management responds most warmly. The engineers' playing is likely to find itself on the priority list for development, marketing and promotion.

And that's about the process through which quadrasonics is passing. It's pretty certain we're going to have to cope with it, whether we like it or not.

The purpose of this article is neither to support quadrasonic reproduction nor to attempt to shoot it down in flames. Our purpose, rather, is to off-set some of the very extravagant statements we have seen and heard with a few down-to-earth (if rather critical) observations.

Nowadays, there are not too many people who will argue against the pro-position that twin-channel stereo provides much more effective sound reproduction than the older single-channel system.

Depending on the recording technique used, twin-channel stereo can reconstitute sound sources at the front left of the listening area and/or the front centre and/or the front right. Alternatively, as well, it can create the impression of a sound source (typically an orchestra) spread smoothly between the loudspeakers. By other means (microphone placement, etc) it can also convey an impression of distance, so that a vocalist may seem quite convincingly to be well out in front of the accompanying orchestra or choral group.

If a twin-channel stereo system does not provide this kind of sound information, it is simply not operating to best advantage.

Unfortunately, it is all too easy to jump to the conclusion that, if two channels are better than one, four channels must automatically be better than two!

What is more, if four channels can create the impression of being at the actual performance, then surely that must be approaching the ultimate in sound reproduction.

In fact, there is good reason why one should not give automatic assent to either of these propositions. They need considerable qualification.

At a live performance, the preferred listening position is usually towards the front of the auditorium.

Why?

Because such a position ensures a generous proportion of direct sound, without too much accoustic clutter from echoes or from noise created by the audience.

Indeed, if it were not for the fact than an audience provides moral support and acoustic damping, conditions would probably be better for an individual listener if the rest of the patrons had stayed home!

And what about the echoes?

If certain sounds are intended to be ponderous, then echoes (or reverberation) might be considered to add usefully to the total "ponder"! But if the sounds are supposed to be delicate, subtle and cleanly defined, echo can very easily become a liability, just like audience noise.

In short, it is making far too sweeping an assumption to claim that addition of auditorium ambience will automatically add to the enjoyment or "listenability" of recorded music. In some cases it might; in others it could as easily detract from the clarity of the direct sound. And there can surely be little justification for transferring into the listening room, at considerable cost, the poor acoustics of an unduly reverberant auditorium, or the distraction of an unduly noisy audience.

If the quadrasonic technique is to contribute to the listenability of music, it must do better than produce an ambience made up from any sort of reverberation and any sort of audience noise. Recording engineers will have to accept the additional task of electronically placing the listener in a suitable seat, in a suitable auditorium, in the company of a considerate audience.

There follows naturally the question as to whether a listener in the home can share this optimum environment with an adequate number of other people in a typical room. If seating is reasonably critical with stereo, it will probably turn out to be even more so with quadrasonic. Those furthest away from the main stereo loudspeakers will not only lose some of the frontal sound but they will be getting more than their share of output from the rear loudspeakers. Acoustically, they will have been transported well back in the auditorium to where the echoes come from!

There is also the question as to how effectively sound alone can build a convincing environment. Maybe it can with the eyes closed but it is certainly less than convincing when the ears say one thing and the eyes say another. Even two-channel stereo suffers from this problem if the two loudspeakers are obvious to the view.

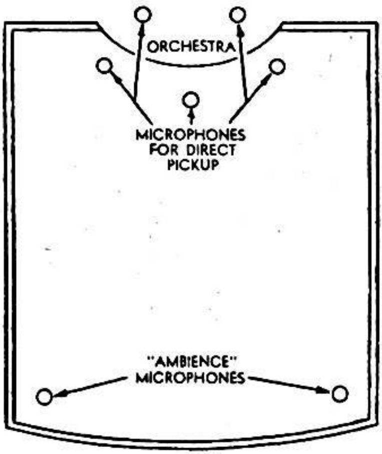
In fact, the part that the eyes play cannot be overlooked. At an original performance, they contribute considerably to the ability of the listener to concentrate on performers or on instrumental groups, tending to relax the demands on the sound itself. Take away the visual, as for audio-only recreation, and the sound from even an optimum seating position may turn out to be lacking. This would tend to undermine the whole "purist" concept of recording and to tip the scales in favour of dispersed or supplemtntary microphones.

In other words, quite apart from the means of recording and reproducing four channels of sound, there is room for a whole lot of discussion about the desirability and problems of so doing, along the lines we have just mentioned.

These problems aside, however, one might speculate as to the microphone set-up which recording engineers might use to capture information for the front and rear loudspeakers of a quadrasonic system.

The "purist" approach would probably favour a group of four unidirectional microphones located in a suitable "listening" position and directed approximately towards the respective corners of the auditorium.

Two, acting as a stereo pair, would capture the direct sound from the source. Their opposite numbers would gather the sound from the rear. At the same time, each pair facing to the right or left would capture sound from the respective sides, as illustrated.



Present-day stereo recordings are commonly mixed down from tracks recorded with multiple microphones. It is reasonable to assume that this kind of technique would be freely used in the production of quadrasonic recordings.

Subsequent reproduction would normally involve four loudspeakers, one in each corner of the listening room.

The two fronting the listening position would give a fairly normal stereo version of the direct sound. The two at the rear would give a stereo version of the reflected sound.

The right-hand front and the right-hand rear loudspeaker would together give a stereo version of sound from the right-hand wall of the original auditorium; the other two would similarly recreate the left-hand environment.

In other words, four loudspeakers, fed with spatially appropriate signals would provide stereo coverage from four directions, re-creating an all-round sound environment.

Bearing in mind present-day stereo recording techniques, quadrasonic recording will not necessarily conform to this purist concept. Engineers will more likely prefer to use multiple microphones adjacent to the sound source and two or more microphones near the back of the auditorium (see diagram).

This is a basically different approach. Rather than being equivalent to the sound heard by a listener in a central position, the master tracks would contain a record of what would be heard by a number of listeners, one in each microphone position. As they do now for two-channel stereo, engineers at the recording console would mix these down into quadrasonic form – highly satisfactory, perhaps, but none the less manipulated.

However, this kind of manipulation, so familiar in present-day stereo recording, would not be the main cause of apprehension for the would-be quadrasonic convert. This would arise from circumstances quite different again.

For many years recording companies have been using multiple microphones to supply signals to four or more parallel tracks on the master tape. These have subsequently been mixed on to a two-track master for the production of normal stereo tapes and discs.

More than once, the statement has been made that, with their huge libraries of multi-track master tapes on hand, recording companies could very readily re-release their catalogues in quadrasonic form.

What this statement ignores is the fact that the vast majority of such tapes contain only information picked up by microphones adjacent to or within the sound source area. No provision has normally been made to record predominantly ambient sound; and what has not been recorded certainly cannot be re-recorded.

What will almost certainly happen is that recording companies will resort, in many cases, to artificial reverberation devices to simulate the echoes for the "ambience" channels. Listeners will not be transported into the original auditorium but into a synthetic building whose "walls" have been conjured up acoustically with the aid of springs, plates and tape loops.

If well done, the result might be acceptable, but it would still be artificial.

As we hinted at the outset, however, there is one important aspect of quadrasonic reproduction which has nothing to do with what we have been talking about: ambience, "being there" and all that.

This aspect – or possibility – is to use the four channels quite independently to reproduce individual artists, instruments or groups of instruments; this at the whim of the composer, arranger or recording engineer.

If a master recording has been made with vocalists and instruments distributed over multiple tracks, the contents can be diverted at will to any or all of the channels of a four-track system simply by manipulating controls on the mixing console.

The instruments of thé orchestra can be distributed around three of the channels, with the vocalist on the fourth, thus effectively putting the listener in the middle of the combination.

The instruments can be swapped around at will or, by progressive manipulation of the sliders on the console, they can seem to move slowly around the room in a procession – if there is any point in so doing.

The idea of manipulating the apparent sound source was the basis of RCA's much publicised "Stereo Action" records, where the producers swapped instrumentalists from side to side and even had them seeming to float across the space between the two stereo loudspeakers. It all looked very clever in the jacket notes but lost out badly as far as the listener was concerned. With systems having poor stereo separation, the effect was not all that noticeable; but even where separation was good, the effect was often lost because listeners didn't remember that the piano should have been here instead of there.

With four channels so involved, the effect will certainly be more obvious but whether it will intrigue listeners sufficiently to make them opt for quadrasonic reproduction remains to be seen.

Turning to the technical aspect, there may well be a vast gap between equipment which has thus far been used to demonstrate quadrasonic reproduction and equipment which might emerge as a standard for domestic use.

Undoubtedly, tape lends itself admirably to the quadrasonic technique but in what format?

- Four tracks straight across quarter-inch tape?

- Two lots of quadruple tracks conforming to eight-track dimensions already in use?

- Reel, cartridge, what speed, what track dimensions, etc.?

- Or will cassettes win the day and, if so, in what track configuration?

Then there is the question of quadrasonic disc recordings which would appear to be within the realm of commercial possibility even in their present dimensions. But the recently announced achievement of recording video on an ultra-fine groove format would make child's play of four-channel audio.

But over and above the intrinsic merits of the various possible systems, it is likely that ultimate preferences will be heavily biased towards compatability with present-day stereo systems.

There would be an obvious advantage in producing quadrasonic tapes or discs which could be played as stereo on existing stereo equipment or as quadrasonic on suitable four-channel equipment. In other words, just as twin-channel stereo came into its own on the back of mono microgroove tracks and tapes, so it would be logical for quadrasonic to take over from stereo by simply adapting the existing formats.

Whether this will happen or not remains to be seen but there is good reason for enthusiasts to be cautious before spending a lot of money on equipment which may end up as some kind of an orphan in terms of the ultimate standards.

Finally, what kind of reactions have been registered by people who have been able to listen to domestic style quadrasonic reproduction?

Firstly, there has been a tremendous amount of interest. This is entirely to be expected because audio enthusiasts are the kind of people who will react this way to innovations in their field.

But while there has been some enthusiasm, there has been a lot of caution too. Larry Klein, writing in "Stereo Review" says: "I have attended a number of public and private demonstrations and have usually come away disappointed. As far as I can tell, the major difficulties arise in the recording rather than in the playback process, though the two are of course interdependent. Despite their having been conducted by engineers with some four-channel experience . . . they simply lacked the 'I am there' quality that I hope to experience with classical four-channel recording".

This kind of reservation is shared by the author, who spent some time listening to equipment.

The equipment itself operated faultlessly but, certainly with the tapes available, no amount of knob juggling seemed to produce that warm, spontaneous feeling that the system was achieving a desirable end result. Set too high, the ambience channels distracted and confused the direct sound. Set low enough to avoid any distraction and they were virtually off!

Yet, while turning them down to zero seemed to leave the direct sound clearer, it also seemed more remote that it had been before.

One could not but be aware also of the importance of the listening position because, on a proportional basis, a relatively small movement towards the ambience sources represented quite a journey in the original auditorium towards the place where the echoes were coming from!

The "Stereo Review" writer had this to say:

". . . at several of the demonstrations I have attended, there seemed to be a single, precisely located (for a given recording) 'correct' seat that was at the intersection of a pair of imaginary lines drawn from the front-left to the rear-right speakers. If one moved away from this X-marked spot, the front-to-rear balance went completely askew"

Reactions like this have produced the caution referred to earlier and confirmed the idea that "ambience" style quadrasonic reproduction involves a good deal more than merely contriving some output from loudspeakers behind the listener. It will involve study, experiment and debate no less intense than has accompanied the emergence of good stereo practice.

Only when things are just right will the listener experience any kind of conviction of "being there".

It was certainly unusual to be surrounded by instruments popping up from all directions, accompanied by a drummer who was seemingly able "to do his thing" from any corner at a moment's notice.

One spontaneous reaction from a lass at the demonstration was: "Isn't it fan . . . . tastic?"

But of course, one's reaction to a startling audio gimmick is not necessarily any guarantee that music will be preferred in that form in the longer term. In real life, it is natural to want to get fairly close to an orchestra but not necessarily inside it!

So there we are, Quadrasonic reproduction is an intriguing idea, a technical challenge and an interesting effect – but whether it will become a commercial success remains to be seen.

4 CHANNEL DISCS

After years of speculation, four-channel or "quadraphonic" discs have made their appearance on the market. If you're so disposed, you can buy the four-channel equipment on which to play them. But, before you decide, let's bring you up to date on the overall situation.

During the 1960s two-channel stereo made a virtually clean sweep of the disc record market. At least, manufacturers offered stereo versions of their regular mono releases. Gradually stereo discs accounted for an increasing share of the market until mono discs became the poor relations.

But technology seldom stands still. During the past decade, while the vast majority of enthusiasts were listening happily to their two-channel stereo systems, a few producers, engineers and manufacturers have been dreaming about additional channels which would allow them to introduce sound sources behind the listening position, as well as in front of it.

Why do this?

To the engineer quadraphonic sound represented a technical challenge.

To the purist musician, it offered the possibility of reproducing the total ambience of the original auditorium, allowing the listener to get a little closer to the feeling of "being there".

To the producer of musical spectaculars it offered the chance of "surround stereo", placing the listener right inside a circle of performers.

To the manufacturer, it held the promise of a whole new market activity, involving more equipment, more amplifiers, more loudspeakers.

It was assumed initially that four-channel sound would have to be recorded and reproduced from tape. Thus the initial engineering effort was aimed at evolving suitable multiple heads and adapting existing stereo tape equipment to record and play four parallel tracks.

And since the tracks could be entirely separate, each as important and distinct as the other, such a concept was commonly described as "discrete" four-channel sound.

As far back as 1961, the Nortronics Company in the USA sought to introduce their "stereo-four" discrete tape system but it was clearly ahead of market demand.

Then, around 1969, there was a new flurry of interest in four-channel tape around the well known names of Vanguard, Acoustic Research and Columbia. There were demonstrations around the world, magazine articles, the announcement of four-channel equipment and tapes but that was about as far as it went.

Commercially, the message seemed crystal clear: consumers were simply not ready to step directly from two channels on disc to four channels on tape.

What was necessary was a method of impressing four channels on disc – but in such a way as to be compatible with present stereo players. Customers could then be expected to migrate in their own good time, from two channels to four – just as they had done from mono to stereo.

But how could four sets of information possibly be impressed on a groove which geometrically seemed capable of responding only to two sets of information? It seemed impossible – or at least impractical.

There was also the question of stereo FM broadcasting. Here was another audience locked to two-channel sound. Somehow, another two channels had to be loaded on to the carrier without compromising existing services and equipment. It seemed like a parallel problem: solve one and you have most likely solved the other.

In 1969, Peter Scheiber, an enthusiast-engineer-musician announced that he had devised a method of compressing four sets of audio information into two channels, which could then be recorded in the ordinary two-channel stereo format.

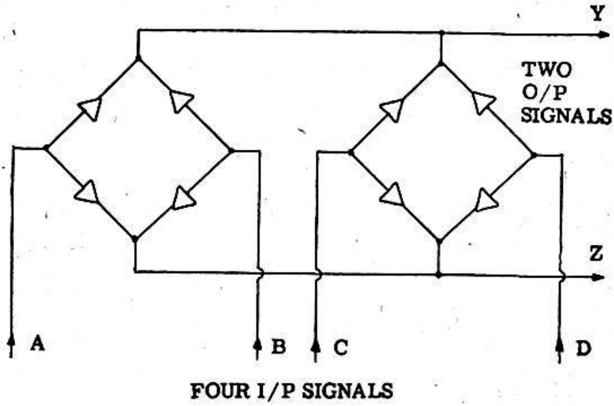


Fig. 1 : A basic 4-2 encoding matrix, from four channels to two channels. The proportions of A, B, C & D which appear on Y and Z depend on the gain and phase characteristics of the individual matrix ampliers.

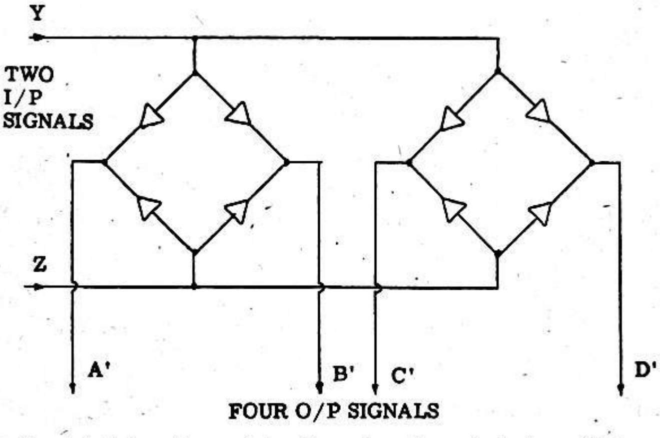


Fig. 2 : A 2-4 decoding matrix, from two channels to four. Note that the output signals are branded as A', B', C' & D' implying that they are not identical with the original input signals A, B, C & D.

He claimed that the recording could be played either as a two-channel disc or, with supplementary decoding equipment, as a four-channel source. Moreover, the same encoding and decoding principle would be applicable to twin-track tape and stereo FM broadcasting.

Industry was cautious at first, even dubious, but demonstrations seemed to establish the validity of his claims. But what of the patent situation, rights, royalties and all that? Here was a challenge for the engineers and management of any number of large hi-fi concerns. Laboratories around the world turned their attention to "4-2-4" audio: four channels into two, then back into four.

One by-product of this concentrated effort was a rash of systems aimed at synthesising a four-channel sound from existing two-channel recordings. The method suggested by David Hafler was the 2-2-4 approach (two initial signals, two channels, 4 outputs) offered a link between existing program material and 4-channel replay equipment but it was commonly viewed as a short-term expedient. In fact, for reasons which will become apparent later, there is more affinity than might first be expected between these 2-2-4 systems and some of the commercial 4-2-4 approaches which have evolved from Scheiber's proposals.

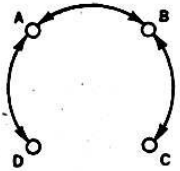


Fig.3: Three independent signals can be impressed on a four-wire system and be recovered without significant cross-talk.

Scheiber's basic approach to the problem of compressing four signals on to two channels is by the use of a matrix network, which is shown in generalised form in Fig.1. It is assumed that a complementary matrix will be provided to recover the four signals from the two-channel medium at the point of replay. More of that later, however,

While it was aimed primarily at solving the problems of disc recording, the idea of matrixing is equally applicable to ordinary stereo tape equipment and stereo FM broadcasting. A matrixed signal contains only audio frequencies and only two channels, and it can be copied and transmitted just like any other two-channel stereo. Only in the listening room does it need to be decoded and reproduced as four channels.

It is interesting to note, in passing, that even a mono input signal via any one channel will tend to place a signal on both Y and Z. A two-channel stereo via any two channels will tend to place two signals each on Y and Z.

Considering the 4-2 (four-to-two) matrix of Fig.1, four input signals A, B, C & D are fed through amplifiers to the two output channels Y & Z. It is for the designer of the matrix to determine what shall be the relative gain of the respective matrix amplifiers and their phase and frequency characteristics.

The options in designing a matrix are almost endless. If, for example, the amplifiers all had the same gain and phase characteristics, equal proportions of all four input signals would appear on each of the two output lines. This would obviously be a rather pointless result, since the four inputs would simply have been mixed into two identical mono channels!

However, by setting up the amplifiers with different orders of gain, certain of the input signals can be made to dominate one or other of the output channels.

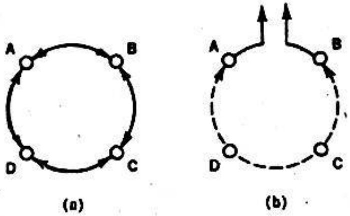


Fig.4: Problems arise when a fourth signal is introduced on to a four-wire system (a). The signal recovered from A-B, as shown in (b), is no longer pure but is polluted by elements of the other three signals.

Again, by reversing the output phase of one or more of the amplifiers, certain input signals can appear on the output lines in opposite phase.

In fact, the designer can do virtually anything he sees fit to any of the matrix amplifiers, in terms of gain, phase, frequency response or dynamic characteristics, to obtain what he judges to be the most desirable end result.

Certain requirements have to be observed, however, if that result is to be compatible with present-day two-channel stereo systems, or with older mono players or for mono broadcasting.

If signal A is intended for the left-front loudspeaker, it should clearly dominate, say, output channel Z so that it will be reproduced in the left-front loudspeaker of an existing two-channel stereo system.

Similarly, if signal B is intended for the right-front loudspeaker, it should clearly dominate output channel Y.

If too large a sample of input A should appear on output Y, or too much of input B should appear on output Z, the apparent separation in ordinary stereo mode would be prejudiced

The influence of signals C and D also has to be considered. If C happens to represent the left-rear signal, it might conceivably be allowed to mix with the left-front on line Z, with D going to line Y. Again, too much of C and D on the opposite output lines could upset normal stereo separation.

Here the matrix designer faces a dilemma. He must allow a fairly generous mix of all four inputs on the two output lines if he is going to achieve his ultimate goal of unscrambling them into four signals. Yet too generous a mix could result in a record with such debased separation in ordinary stereo mode that it would not be acceptable to customers who have their own ideas as to how a two-channel record should sound!

Hence the idea of playing tricks with the phase of the mixed signals. By putting the "unwanted" component on the respective channels in reverse phase, the subjuctive effect may be to increase the apparent separation. The engineer has the option of putting either or both B and D on line Z in opposite phase, setting the level so that the subjective separation in two-channel stereo mode is adequate.

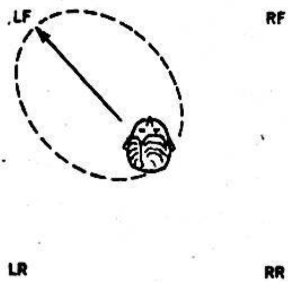


Fig.5: If a signal intended only for the left-front loudspeaker (LF) is reproduced at lower level in loudspeaker RF and LR, the sense of direction, represented by the arrow, becomes broad and vague (ellipse) to a degree governed by the amount of cross-talk.

But there is still another important consideration: what happens when output lines Y and Z are paralleled as for mono reproduction? Components which are equal and out of phase on the two lines will cancel and virtually disappear. The end result could be that the total output signal is lowered to an unacceptable level by cancellation. Worse still, vital musical information might virtually disappear.

Along with this is the uncomfortable possibility that incomplete concellation, the result of phase aberration, could leave remnants of the cancelled signals to be interpreted as odd sounds or straight-out distortion.

To sum up what has been said, a recording engineer, in aiming for an ultimate four-channel recording, has to make sure that the matrixing will provide a completely acceptable result, when played in normal stereo and mono mode. If this is not achieved, the recording will lose its appeal for the existing mass player market and for mono broadcasting, defeating the whole concept of compatibility.

These requirements notwithstanding, the ultimate objective is, of course, to produce a two-channel signal which can be decoded into four signals, hopefully equivalent to the original A, B, C & D.

This involves a matrix network as in Figure 2, being the converse of Fig.1. The two input signals are fed into Y and Z and emerge as signals A. B, C & D.

At this point, the reader can be excused for demanding a re-statement and verification. It is one thing to mix signals but how can they be separated out again if they have not been "tagged" in some way in terms of frequency (as by a carrier system) or time (in terms of time multiplexing)?

In fact, it would be quite a task to analyse the operation of a decode matrix in so many words, and without resource to copious and tedious algebra. However, it may be helpful, in resolving the difficulty, to realise that, by proper segregation of wiring, two channels can provide the equivalent of a four-wire AC circuit: four wires into the record head (tape or disc) and four from the replay head.

With four wires available there is no special problem in conveying three separate signals (between D-A, A-B, B-C) as in Fig.3. Most likely, three phases of the ordinary power mains are carried into your home on four wires for independent use. There is certainly nothing new about this and, if the record manufacturers had been content to settle for a tri-phonic system, further difficulties might largely have been avoided.

But they weren't content and, to match the challenge of the tape medium, they insisted on introducing a fourth signal. In terms of a basic four-wire circuit, it involves introducing a signal between C and D, as in Fig.4a.

This closes the loop and, fairly obviously, some of the new signal voltage between C and D will be distributed across the other paths D-A, A-B and B-C, in proportions depending on the relative impedance levels.

By exactly the same reasoning the voltage between B and C will distribute around C-D, D-A and A-B. The same will be true of the remaining channels.

Because of this interaction, it will no longer be possible to pick off from A-B a simple signal, as would have been the case for Fig.3. As indicated in 4b, the signal across A-B must contain a proportion of the signals B-C, C-D and D-A. In turn, the signals across each of these other channels must be similarly "polluted"

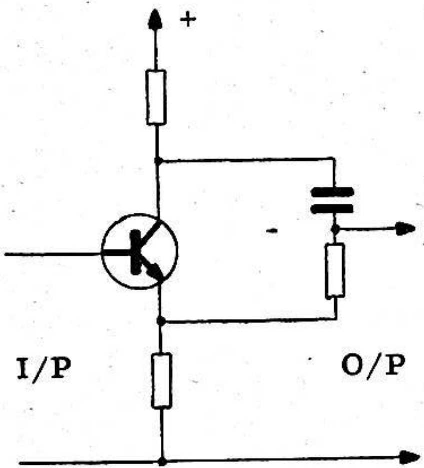


Fig. 6 : This type of circuit can rotate the phase of an audio signal by 90 degrees at a particular frequency and by differing amounts at other frequencies

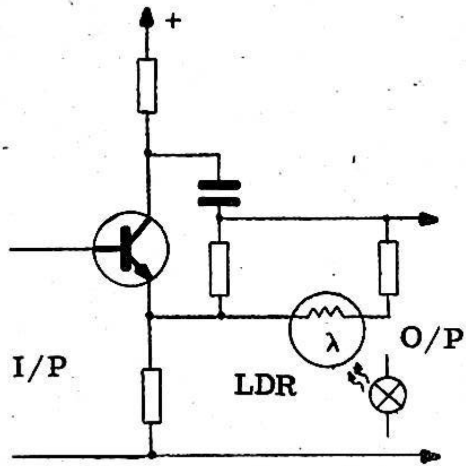


Fig. 7 : This phase modulator circuit, well known to electronic organ enthusiasts, has made its appearance in the context of quadraphonic matrixing!

It is for this reason that the outputs from the decode matrix are designated as A′, B′, C′ & D′. The implication is that they may relate to, but will differ from A, B, C & D.

What would seem to emerge from this is that a two-channel (four wire) system can carry, at most, only three discrete (or separate) signals. The penalty of trying to handle four signals is a high order of crosstalk.

Not satisfied with mere words and generalities, the Author took it on himself, while this article was being written, actually to work out typical figures, starting with the original Scheiber matrix. Several sheets of paper later, he was convinced that elements of the original signals could in fact be recovered from a 2-4 matrix, but only with a significant degree of cross-talk which, in some cases, could be within 3dB of the wanted signal!

How small that figure sounds to enthusiasts who are apt to frown if the crosstalk in a stereo pickup falls below 20dB!

As we have indicated, the encode and decode matrices can be manipulated within wide limits, and the phase of the signals varied to achieve different orders of cross-talk between different channels. However, measures to decrease cross-talk between certain channels are likely to increase it between others.

No one has yet succeeded in developing a 2-4 matrix which is free from this problem, despite impressions to the contrary which might be created by euphemistic articles and advertisements.

Proponents of the matrix system admit to the cross-talk problem but claim that it is not nearly as serious, subjectively, as the figures might suggest. Frontal left-right separation has to be preserved but they claim that a quite small loudness difference between the front and rear channels on the respective sides is adequate.

From this point on, the literature is full of ideas for "processing" the respective signals in order to achieve the most acceptable end result. Nor is it immediately clear as to how many of these ideas are theoretical proposals and how many are actually incorporated in existing or emergent matrix systems.

One broad approach, which seems to have been dignified by such terms as "logic" and "digital", is virtually an expandor system. In-built circuitry senses signals which are unique to particular channels and seeks to emphasise those signals by a dynamic change in gain of the appropriate amplifier path.

Another approach, as already mentioned, is to reverse the phase of certain signals in certain paths, so that they tend to cancel rather than add. By this means, for example, a signal intended for the respective front loudspeakers may be eliminated specifically from the diagonally opposite rear loudspeaker, though it will still be present as cross-talk in the right front and left rear.

Such cross-talk tends to broaden the apparent sound source for the particular signal, as illustrated in Figure 5. While this may not be important to a listener in a near optimum position, it will lead to problems for listeners elsewhere in the room. As the listener moves towards right-front or left-rear, they will hear more of the intended signal from that particular loudspeaker. The apparent sound source will move towards that loudspeaker and might ultimately be identified with it.

In fact, it is possible to derive a whole series of apparent sound source patterns for typical matrix networks and, in particular, those which satisfy the basic requirements stated earlier for quadra/stereo/mono compatibility. For good measure, these can be expanded to take in what is claimed to be a major difference in the ability of listeners to discern sound sources in front of them and behind them.

What emerges from this kind of analysis is that a matrix system can be devised readily enough which will satisfy the basic mono/stereo/quadra requirement for a strong cente-font sound image and acceptable left/right frontal spread.

But having satisfied this basic requirement, it is difficult to produce a firm, isolated image from the respective rear loudspeakers and even more difficult to create an even sound spread across the rear wall. There is a strong tendency for sounds intended for the centre rear to be interpreted as coming from centre front.

The technique, which was mentioned earlier, of reversing the phase of some signal components has a complex effect. Electrically, it can produce actual cancellation of signal components to diminish the severity of particularly troublesome cross-talk.

Acoustically, there is the possibility that a deliberate anti-phase component may tend to confirm a listener's judgment that a particular sound is NOT coming from a particular direction. Fairly obviously, this kind of thinking cannot be pushed too far, because phase tends to become random once sounds have been projected into the listening room.

However it is obvious that, having manipulated matrix constants, gain and phase reversals to the limit, some engineers have still not been satisfied with the end result. They have accordingly resorted to other tricks of electronic circuitry. One of these is illustrated in basic form in Fig.6.

It involves a transistor amplifier operating into a split load such that the signals at collector and emitter are of equal amplitude but opposite phase. The actual output is taken from the junction of an R/C network between the two.

Depending on the choice of constants, the output can be arranged to exhibit a 90-degree phase shift at some particular frequency in the audio range. At higher frequencies, the phase will approach that of the collector; at lower frequencies it will approach that of the emitter.

By upending the R/C network, the phase would rotate in the opposite manner.

Networks like this may, for example, be introduced into the signal path for the rear loudspeakers if the intention is to blur their sound image and create a more effective feeling of ambience.

But that is not the end.

In Figure 7, an Light Dependant Resistor has been introduced so that the phase of the output signal can be subjected to external control. Electronic organ fans will recognise the circuit immediately as the basis of electronic phase modulation, as applied to organs and other such instruments.

With such a circuit, the phase of selected channels can be rotated in response to an external cyclic signal (as per the electronic chorale in an organ) or it can be modulated by a voltage derived from the signal itself.

This statement, along with Figures 6 and 7 take the mystique out of drawings which refer to spiral groove modulation. By shifting the phase of certain drive signals to the cutter by 90 degrees a certain "radial" or "spiral" quality may be given to the groove but does not obviate the crosstalk problem. In the ultimate, the cutter and the replay stylus know only two vectors and signal current can only be translated into and out of these two vectors. This simply means two channels nothing more and nothing less.

What is apparent from all this is the enormous range of options open to recording engineers, and the opportunities for as many "standards" as there are studios. What chance has the record buyer of identifying the original encode matrix used and selecting the appropriate one for decoding?

In fact, it would not seem out of place to ask whether, in certain cases, there is such a thing as "an appropriate one for decoding". From within the industry, critics of the whole approach claim that it is rapidly moving into the area of electronic processing, with the result supposedly justifying the means.

There is talk overseas of switchable replay matrices and claims of universal matrices which give an acceptable result with all present matrixed quadraphonic records. But, right now, the vagueness threatens to be of far greater magnitude than it was twenty years back, in relation to compensation curves!

There is this difference, however: in the matter of compensation curves the enthusiast was matching one curve with another in the reasonable hope of obtaining a fairly accurate end result. With matrixed quadraphonic, the intangibles are far more numerous and the end result can only be a compromise which has all the built-in limitations of having tried to cram in one channel too many.

Subjectively, the end result may be acceptable, even very pleasant, but, at the grass roots, the whole structure of compromise and subjective reaction is at cross purposes with the true aims of high fidelity reproduction.

Maybe it is progress, but in a rather oblique direction!

And here, rather curiously, we find ourselves almost back to something referred to earlier in this article: to methods of processing existing stereo recordings to extract and exploit rear ambience and other effects. Basically, these depend on extracting the difference between the left-front and fight-front stereo signals and reproducing this difference at the back of the room from out-of-phase loudspeakers.

It can be done quite simply, as per our earlier reference, using a couple of extra loudspeakers.

Alternatively, the L and R signals can be fed into a supplementary combining unit, allowing them to be "fiddled" to a greater or lesser extent, using some of the techniques already referred to. This done, they can be passed to a supplementary stereo amplifier driving a pair of loudspeakers at the back of the room.

It may well be that, in many cases, the end result will not be all that different from that obtained using matrix techniques.

Simulated quadrophonic should therefore not be written off as a superseded gimmick. It is a significant and still useful step in the direction of matrixed quadraphonic.

But matrixed quadraphonic is not the ultimate. It has definite limitations and relies significantly on simulation.

What of the future?

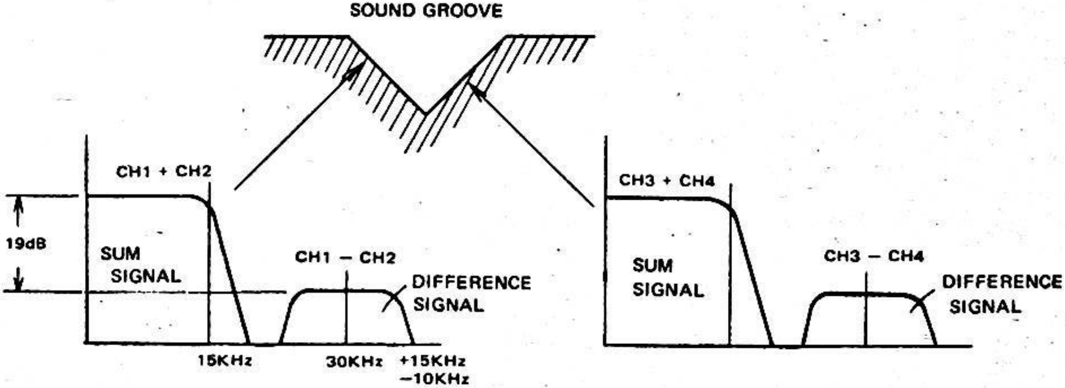
Well, matrixed quadraphonic may meet industry and user needs for the foreseeable future. It certainly has a lot going for it:

The records are here now and they are fully compatible with present-day players. The kind of decoding equipment which is necessary to process them can also be used to simulate quadraphonic sound from present discs and even expand mono discs where this seems desirable.

Matrixed quadraphonic is equally applicable to twin-channel stereo tape machines and Stereo/FM broadcasting. And it has the backing of numerous record and equipment manufacturers.

But . . . and it is a very significant but.

It is not a "discrete" system. The four channels are not genuinely independent. If the market ultimately demands that they should be, the matrixed system will have to give way to something else – either four tracks on tape, or the JVC/Panasonic/RCA multiplex system on disc, involving a superimposed carrier.



: The CD-4 system uses the basic geometry of two-channel stereo but records signals up to 45kHz on each groove wall. This allows two distinct signals to be virtually "stacked" on each wall, one recorded directly at audio frequencies, the other modulated on to a 30kHz carrier, as shown at left.

Commercially, the matrix system has a lot going for it. Being a 4-2-4 system, it is equally applicable to the two-channel disc system, a two-channel tape system or to FM-stereo broadcasting. The original four-channel material can be encoded on to two channels, then replayed, dubbed or broadcast as such, and finally decoded and played back in the home, ostensibly as four-channel sound.

The frequency components involved do not fall outside the existing audio pass band, so that no special problems of compatibility are involved. Matrix-system discs can be played back with existing two-channel stereo equipment, the signal fed to a decoding unit and extra signals made available for amplifiers driving the rear loudspeakers.

Ostensibly, the matrix system offers a complete answer to the problem of obsolesence. Enthusiasts can buy matrixed quadraphonic records and play them for as long as they like on existing two-channel stereo equipment, without risk of damaging the grooves. At some later date, a decoder and additional amplifier channels can be added, and advantage taken of the quadraphonic content of the records.

At the same time the decoder provides the facility to sythesise extra signals from existing two-channel material, so that the enthusiast can gain an additional dimension from older recordings.

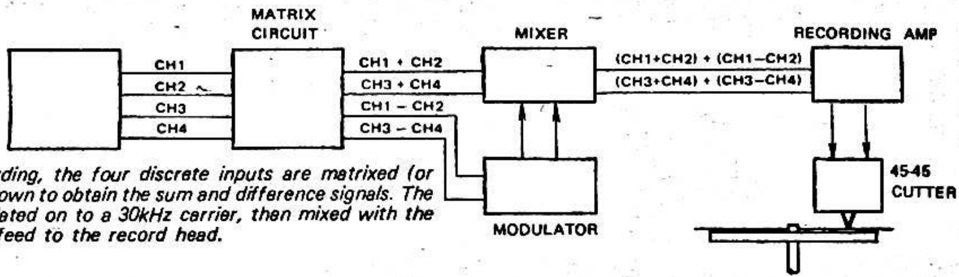
Reflecting the commercial attractiveness of the matrix system, it has no lack of support, at least in broad principle. To quote from a recent Japanese brochure:

"Almost all of the four-channel stereo systems available on the market today are of a matrix system . . . these are listed here for reference purposes . . . .

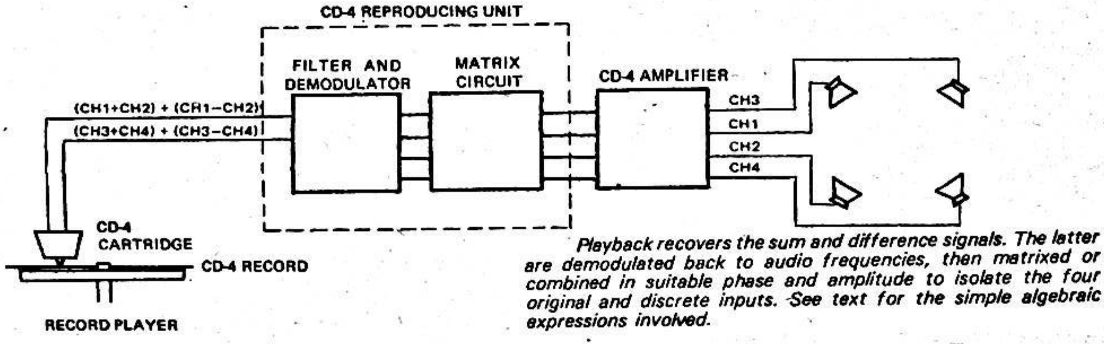
"Toshiba	QM system
Denon	QX4 system
Matsushita	AFD system
Sansui	QS system
Kenwood	QR system
Hitachi	Ambiphonic
Sanyo	QSC system
Mitsubishi	QM system
Onkyo	X–1 model
SONY	SQ system
TEAC	A–2400 model
Pioneer	Quadrilizer"
JVC/Nivico	SFCS.

This list does not include manufacturers in countries other than Japan, nor does it reflect the backing for the system from companies marketing complementary discs and tapes.

But, as we pointed out last month, for all its convenience and attraction, the matrix system falls short in one vital area: it is not a true four-channel system nor, presumably, can it ever be.



For recording, the four discrete inputs are matrixed (or combined) as shown to obtain the sum and difference signals. The latter are modulated on to a 30kHz carrier, then mixed with the sum signals to feed to the record head.



Playback recovers the sum and difference signals. The latter are demodulated back to audio frequencies, then matrixed or combined in suitable phase and amplitude to isolate the four original and discrete inputs. See text for the simple algebraic expressions involved.

It suffers intrinsically from cross-talk between channels, and individual companies have sought to offset this limitation with all manner of electronic processing. Whatever the final decoding system recovers, it certainly cannot recover four original and discrete channels from a two-channel medium.

While the limitations of the matrix system will obviously have been well known to recording engineers, the implications of their mathematics and their rather obscure circuitry have not been readily apparent to the majority of technical writers. The loudest message has been the commercially inspired one that the four-channel disc problem had been solved by the matrix system (albeit rather mysteriously) and that it was all over bar the shouting!

Gradually, however, the contrary opinion has filtered through, along the lines expressed in our last issue – though not in anything like as much detail.

A recent issue of J.E.I. (Japan Electronic Industry magazine) carries an article headed: "4-Ch. Stereo Systems Pushed Strongly, But Lack True Definition, Development".

Discussing the subject, the writer says:

"Among the records already on the market, those with more echo components and recordings of actual performances may well be called the matrix records, because most matrix records are more befittingly described as variations of two channel records, rather than four-channel records".

One of the companies which has taken a strong contrary line in the four-channel arena is JVC/Nivico – JVC standing for Japan Victor Company. In a recent publication, one of their writers says:

"Separation is incomplete in a matrix system. Thus a 4-channel record is not very different from a prior stereo record. It is advantageous from the standpoint that a conventional stylus and cartridge can be used without modification. However, it has a problem in the complete separation of the four sounds, which is the most important requirement for 4-channel stereo systems. It is not possible by the matrix system to pick up one sound alone."

JVC has, in fact, done the lion's share of research into systems which hold real promose of a true four-channel capability – as expressed in the capacity of a system to produce sound from any one of four loudspeakers in isolation, or from any number of those loudspeakers in any desired proportions.

With four-track tapes it is no great problem. With discs it is a problem because, fundamentally, a stylus can only respond reasonably to two vector forces, displaced from each other by 90 degrees. It is a question of making those two vectors do four jobs.

The system finally adopted by JVC assumes the use of a normal stereo groove, with each wall at 45 degrees from the horizontal. However, instead of each wall carrying just one audio signal (right channel or left channel) each carries two distinct signals, impressed simultaneously by the recording cutter. But unlike the matrix system, they do not share the same frequency band.

One of the signals is at audio frequencies in the range nominally between 30 and 15.000Hz.

The other, having first been frequency modulated on to a 30kHz carrier, occupies a range of frequencies between 20,000Hz and 45,000Hz – i.e. 20 to 45 kHz. To invoke an old PMG term, they are "stacked" in terms of frequency.

Each wall of the groove thus carries a complex pattern of frequencies ranging from about 30 Hz to about 45 kHz representing the content of two separate and distinct audio signals. Between them, the two groove walls carry information about four separate audio signals. (See Figures 1 and 2.)

The playback cartridge can be designed along broadly conventional lines but it must be capable of responding to this very wide frequency range without prominent peaks, troughs or resonance effects. In broad terms the frequency capabilities need to be about two-to-one up on existing high quality stereo cartridges.

The complex pattern of frequencies recovered by each half of the cartridge is fed to a frequency dividing network. (Figures 3 and 4.)

Frequencies in the range 30 Hz to 15 kHz are separated out, to become one of the signals originally fed to the corresponding coil of the recording cutter.

Frequencies in the range 20 kHz to 45 kHz are likewise separated out, fed to an FM demodulator, and thus used to recover the second signal fed to that cutter coil.

From the two groove walls and from the respective halves of the cartridge, four separate signals are thus obtained.

At first glance, one might assume that the stereo signals for the front loudspeakers would be recorded on the respective walls as the basic audio component. Further, that the stereo pair for the rear loudspeakers would be impressed on the 30 kHz carriers. But in fact, for a variety of reasons, JVC have chosen not to do it this way.

Instead, they matrix (or combine) channel 1 and channel 2 together and inscribe the resultant "sum" signal as the basic audio pattern on one wall of the groove. Channel 3 and channel 4 are inscribed on the other wall. The logic of this approach is not hard to discover.

If we assume that channel 1 is front left and channel 2 is rear left, the sum of the two (Ch.1 + Ch.2) representing the total left signal ends up as the basic audio pattern on one wall of the groove.

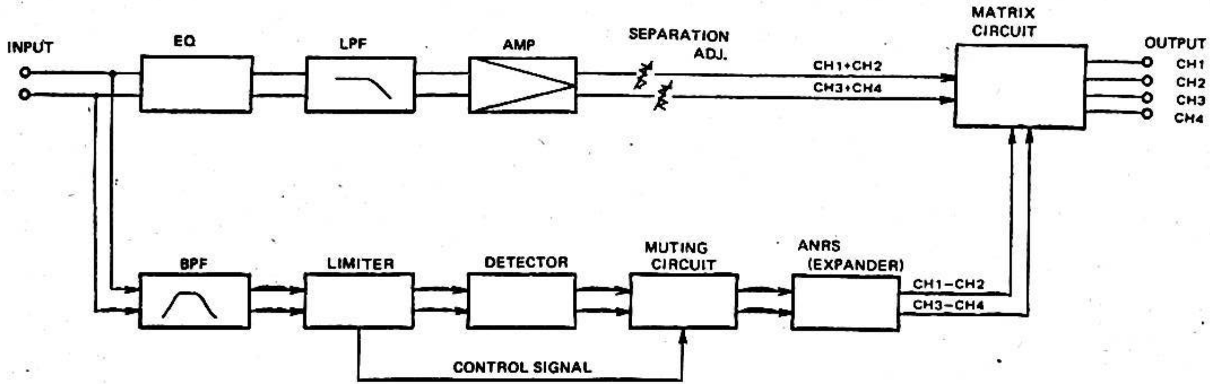


Fig. 4 : The essential elements of a CD-4 demodulator unit. Output from the pickup is fed into the input terminals, left. Audio components, representing the sum signal pass through a low pass filter (LPF) while the difference signals have to be separately demodulated and processed. They are finally combined in a matrix to produce 4-channel discrete channels.

Similarly, the total right-hand signal ends up as the basic pattern on the other wall of the groove.

If played on a two-channel stereo system, the disc is heard as a two-channel stereo disc, with normal separation between left and right but, of course, with front and back combined.

If played in mono mode, the components add again, to produce a normal mono signal.

What of the high frequency components, which are also inscribed in the groove walls?

Very simple.

The majority of styli and cartridges will not respond to them very effectively, and they will be further attenuated by the normal de-emphasis and tone control circuitry. What is left, still has to get through the loudspeaker. Last but not least, frequencies about 20 kHz are outside the range of hearing anyway!

In this respect, therefore, the discs can be accepted as playable on, and compatible with, existing stereo and mono equipment.

What is actually modulated on to the two high frequency carriers is the "difference" between the respective pairs of signals. The same groove wall which carries (Ch.1 + Ch.2) as a direct audio signal, also carries (Ch.1 – Ch.2) modulated on to its 30 kHz carrier.

In four-channel mode, it is necessary to recover and demodulate the 20 kHz to 45 kHz components to isolate the audio difference signal (Ch.1 – Ch.2). Then by adding samples of the sum and difference signals in suitable amplitude and phase, the individual components can be recovered. The algebra is very simple:

$$(Ch.1 + Ch.2) + (Ch.1 - Ch.2) = 2 \times Ch.1$$

Again:

$$(Ch.1 + Ch.2) - (Ch.1 - Ch.2) = 2 \times Ch.2.$$

In short, the Channel 1 signal and channel 2 signal can be recovered, substantially in their original form.

The same applies to channels 3 and 4.

Because it is theoretically possible by these means to record and recover four completely separate audio signals, JVC have called their system "CD-4" standing for "Compatible Discrete 4-Channel".

The various steps in the CD-4 system are illustrated in the accompanying diagrams, which should be studied together with the explanatory captions.

While the foregoing sets out the basic principles of the CD-4 system, a perusal of JVC literature points up numerous refinements in detail and approach which represent the difference, no doubt, between a basic concept and a commercially acceptable end result.

A single master oscillator is used as the high frequency source. Its output is split and each signal passed separately through a "Serrasoid Frequency Modulator", for the respective groove walls. This and other precautions in the modulation and demodulation process are aimed at minimising the generation of spurious beats between the respective high frequency signals.

Special attention is paid, not just to the deviations of the recorded groove, but to the path which is likely to be traced by a spherical-shouldered playback stylus. This must be related in turn to amplitude and frequency, and also to wavelength, as affected by the diameter of the particular groove.

To this end, the signals on the master tape are read by separate heads just ahead of the heads which feed the recording stylus. These pre-record signals are analysed from instant to instant, correlated with the groove diameter and used to modify dynamically the input to the cutter. It is, in fact, an extension of the long established JVC/RCA Dynagroove technique.

JVC stress that it is necessary to minimise tracing aberrations, both to minimise distortion as such and to preserve optimum phase relationships in the high frequency modulation components. Also at stake is the matter of intermodulation and cross-talk which can be deteriorated by non-linearities in the system.

JVC diagrams indicate the use of FM pre-emphasis, compression and expansion, and muting – all ostensibly aimed at achieving the highest possible signal/noise ratio.

Interestingly enough, the master disc is cut at less than half speed, with master tape speed and master oscillator frequency scaled down in proportion. This is regarded as an interim technique, however.

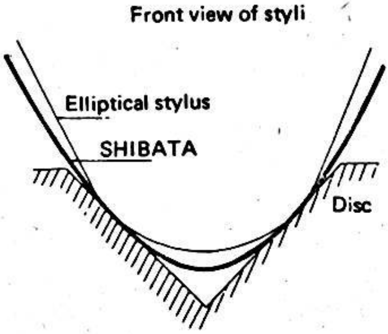


Fig. 5: Viewed from the front, the shoulders of the Shibata stylus have a larger radius of curvature than the conventional elliptical or bi-radial type. Pressure per unit area is reduced, as also is stylus and groove wear.

In all something like 90 patents have been taken out on various aspects of the system.

Draft standards have been presented to the Japan Record Association, to the EIA and RIAA in the USA, and to the European DIN Standard Committee, with a view to encouraging the adoption of the CD-4 system as a world standard.

JVC specifications claim that it is applicable to 12in, 33rpm discs and 7in 45rpm, if need be. Frequency response of each channel is claimed to be 30Hz to 15kHz, cross-talk between channels better than 25dB, and signal/noise ratio better than 50dB. These figures apply to what is on the disc and to be realised in practice, assume the use of a suitably high quality stylus, cartridge and demodulator/decoder.

While Panasonic/National and RCA gave the CD-4 system their formal blessing and cooperation, JVC/Nivico was the first to move it into the commercial sphere with the release of about fifteen albums in mid 1971 straddling the range from rock to classical.

At a press conference in New York, about the same time, the President of RCA Records, Rocco Laginestra, acknowledged "phenomenal progress" during the preceding few months and indicated that RCA was involved in concentrated research which should lead up to its own launch in the near future.

Panasonic was in much the same situation.

In fact, the RCA marketing effort is now rolling and, by the time this issue is in the hands of readers, four-channel discrete records carrying the RCA label should be on sale in American record shops.

RCA's marketing ultimate plan is to release new records only in the CD-4 format, thereby eliminating the need for double stock inventories. They would be played in mono, two-channel stereo or 4-channel stereo, according to the buyer's own equipment.

Despite the confidence and influence of the JVC/RCA/Pansonic group, the CD-4 type of disc yet has to prove its commercial superiority over the simpler, though less ambitious matrix type.

In two areas at least, it faces an obvious disadvantage. CD-4 discs cannot, as yet, be broadcast directly over stereo/FM stations, because the frequency content exceeds what can be contained in the authorised spectrum.

Again, the CD-4 signal cannot be handled by ordinary audio circuits or dubbed for ordinary 2-channel tape replay, because of its 45kHz bandwith.

Of more immediate importance to high fidelity enthusiasts is the durability of the high frequency signals inscribed in the groove.

How many times can CD-4 grooves be played with suitable equipment before the fine serrations become noticeably degraded?

What will happen to a CD-4 disc if it is played even once with a too-heavy, not-very-compliant cartridge? Will the vital difference signal simply be obliterated?

JVC state that research has shown that the average LP disc is played about 20 to 30 times. They appear to be confident that their current production will meet this requirement easily enough, provided they are played with a suitable cartridge and at a playing weight no greater than 2 grams.

They stress, however, that the records can be damaged by older and heavier cartridges. The "sum" signals would remain as normal stereo, but the "difference" signals, necessary to re-create the rear channels, would be at hazard.

One of the reasons for RCA's hesitancy was reportedly their need to be assured that the records would be good for at least 100 playings under proper conditions, and less liable to damage in other circumstances.

RCA's answer seems largely to be in the choice of a new and harder grade of vinyl. Supply and processing problems had to be straightened out but the new vinyl is now said to be giving much harder pressings with lower noise than the standard item.

When teamed with a new decoder developed by Lou Dorren of Quadracast Systems Inc, of San Mateo, California, the new records are credited with adequate difference signals even beyond 100 plays.

And, finally, JVC research into the CD-4 technique has produced a new type of stylus, which is claimed to represent a notable improvement on the current elliptical or bi-radial types.

In these conventional types of stylus the combination of the two effective radii produces a minimum area of contact between the stylus shoulders and the groove walls. However, this produces wall deformation which can exceed the elastic limit of the vinyl with playing weights in excess of 2 grams.

In addition, it is claimed that the depth of penetration of the shoulders into the wall modifies the mechanical impedance of the system and makes it that much more difficult to achieve extended frequency response.

The new "Shibata" stylus still has small radius shoulders, in order to trace more effecitvely the smallest wavelengths in the groove wall. However, viewed from the front, the Shibata stylus is more pyramidal in shape, with a larger effective curvature of the surfaces resting against the walls parallel to the modulation (Fig.5). JVC claim that the effective area in contact with the groove wall is multiplied by four times, resulting in less deformation and much lower wear of both record and stylus.

In addition, they claim a marked improvement in frequency response and a reduction in cross-talk characteristics.

While the Shibata stylus is aimed at solving problems in the region above about 15kHz, it will be interesting to see whether it will have an impact on the ordinary two-track stereo market.

But, of course, the big question is not the Shibata stylus. It is the CD-4 system itself. Will the hi-fi fraternity insist on true four-channel capability or will it settle for something somewhat less pretentious? The matrix system, for example?

HI-FI SYSTEMS AND 78 R.P.M. RECORDS

From an environment of microgrooves, stereo records and tape, one occasionally succumb to the urge to get out old 78 r.p.m. records and listen to the music they contain. As often as not, reaction is one of dismay and a conviction that they must surely have sounded better than that.

In approaching this matter, it is essential first of all, to stress the degree to which evolving standards of reproduction have changed our present evaluation of what is good and not good.

In the days when we bought - and obviously enjoyed - 78 r.p.m. records, high fidelity reproduction as we know it now, existed, for most people, in theory only.

They could but speculate as to how their recordings would sound without the persistent crackle from the shellac pressing; their most accessible approximation was the sound from film, heard in the local cinema.

But neither record nor cinema could boast extended high frequency response and this was another area for speculation about what might be. As for stereo and "big" sound, the most obvious source was the Wurlitzer organ or the few theatres which could show Disney's then way-out "Fantasia".

Inevitably, these somewhat detached sonic experiences tended to make one hope for better things from the records of the day and I well recall the unequal struggle to win more convincing sound from them; an effort in which every extenstion of the high frequency response brought with it an unwanted heritage of noise and distortion.

I recall, too, the stimulation which followed the release of Decca's ffrr 78s and first long-playing microgrooves; how they emphasised the shortcomings of earlier discs while still falling short themselves, of the sought for ideal.

What we knew then, and were conscious of, is very much more apparent to ears now grown accustomed to higher standards of reproduction: 78 r.p.m. pressings ARE noisy, ARE limited in frequency range and DO contain a high potential for distortion on replay.

If reproduced on a wide-range system, they are almost certain to sound noisy and distorted; if limited, in terms of frequency response, to minimise noise and distortion, they will sound so dull as perhaps to be uninteresting. Broadly speaking, there is no way out of this dilemma.

Then, too, musical styles and recording techniques have changed drastically during the past thirty years. Vocalists were pushed very close to mic. in those days, to keep them clear of the play back noise level; orchestral accompaniments, themselves in the background, followed stylised oom-pah-pah arrangements; the studios were as dead acoustically as acres of caneite and curtains could make them; dynamic range had to be severely limited.

These are just extra problems which have to be faced, in trying to derive enjoyment from obsolete discs.
In fact, the best reproduction to be heard from pre-war recording efforts is almost certainly via modern re-release on microgroove albums. Frequency, these are derived by playing the original metal "mother" with a super quality pickup fitted with a stylus critically selected to fit the groove. By so doing, distortion is minimised, frequency response is preserved and the inevitable crackles of a shellac pressing avoided.
Where the effort is warranted, recording engineers also have the option nowadays of discreetly modifying response contours and dynamic range, introducing sharp-cut filters and synthetic reverberation, and frequency-dividing for simulated stereo. And, while we may not always agree with their judgment in such matters, it is equally true that they have often come up with re-recorded efforts far more listenable than well-preserved shellac pressings of the same item, bought thirty years ago.
But, assuming that we have a collection of well preserved 78s and want to play them, what are the problems?
Perhaps the first one has to do with stylus fit. Groove dimensions of 78r.p.m. discs varied widely according to their age, their origin and the precise condition of the stylus with which they were cut. In the days of steel and thorn needles, the differences were taken up by the needle wearing itself to the shape of the groove - with an element of vice versa - of course!
With the introduction of sapphire styli with their potential for better reproduction, came the problem of what radius actually to grind the tip. Too small and it would skate around the bottom of the groove, generating serious tracing distortion; too large and it would ride the shoulders, among the surface scratches; just right and it would ride the groove walls as intended.
The uncertainties of stylus fit were such as to warrant the Goldring organisation marketing the "Headmaster" pickup, with interchangeable heads fitted with different styli. As we recall, there was a 0.0035in stylus for grooves having a freely radiused bottom, a 0.0025 in general purpose stylus and a 0.002 in for those with a more sharply defined "V" bottom.
The listener had the option of using the large radius head, where skating seemed to be a problem and the small radius head where it was not, the latter giving better high frequency response because of its ability to trace finer undulations.
By and large, the days of trying to win good reproduction from old 78 r.p.m. discs have gone and, with them, the market for such specialised pickups. Unless the present-day enthusiast is prepared to go to more than the usual amount of trouble, he will merely purchase a "78" type head with a stylus having a nominal radius of, probably, 0.0025in.
And herein lies the possibility of different behaviour with different records and systems. According to the way an individual stylus, of somewhat nominal shape and diameter, happens to fit an individual groove so will it get down into the skating zone, or up into the noise zone, or say somewhere safely in between.
That is, of course, providing it does not happen to ride on a previously damaged region of groove wall!
The resonance characteristics of the cartridge will also have an impor-

tant bearing on the reproduced sound. The peak may be well suppressed or it may be prominent and so placed as to coincide with a prominence in the loudspeaker, thereby heavily aggravating both distortion and noise.

And here a word about scratch filters - the time honoured subject of many questions:

In the early days of magnetic pickups - the so-called "rock crushers" the resonance usually came out in the vicinity of 3,500Hz, the result of a fairly standardised kind of needle, holder, armature and suspension. As a result, this was the frequency at which scratch tended to be most prominent, and there was good sense in providing an absorption filter, broadly tuned to this region, to counter the peak.

Unfortunately, the practice led to the idea that scratch had a particular frequency of its own and enthusiasts ever since have never quite left behind the impression that they can tune it out. In fact, scratch has no natural period of its own. All that can be said is that it is most prominent at that frequency at which any part of the system happens to be resonant; the object should be to attack any such resonance, or avoid it in the first place.

Incidentally, the "scratch filters" fitted to many present-day hi-fi amplifiers are little more than top-cut circuits, involving a switch and a couple of capacitors. They are in no sense resonant, merely rolling off the treble response in much the same way as would be achieved by turning down the treble control.

Whatever the success one might have in finding a stylus to suit the groove and a cartridge free from troublesome resonance, one of the greatest enemies of "acceptable" reproduction from old records is an amplifier system ending up in a couple of modern wide-range loudspeakers.

While the record itself may not be able to produce actual musical content above about 6000Hz, the wide-range system will see to it that every vestige of distortion and surface noise is reproduced to full advantage!

Nor does it necessarily correct matters to turn down the treble control. While this control might seem quite drastic in its effect on modern, wide-range reproduction, the usual 6dB/octave slope is actually quite modest and its ultimate effect on the audibility of high frequency energy less drastic than one might imagine.

The validity of this statement can be tested by noting how poorly the ordinary treble control copes with a 10KHz heterodyne between adjacent radio stations. As often as not, the heterodyne is still disturbingly audible with the treble control turned fully off.

In the heyday of 78 r.p.m. records, the problem seldom arose because the loudspeakers of the day provided their own in-built treble roll-off with a slope much steeper than 6dB per octave! This was often compounded by an output transformer having high leakage reactance and no feedback or else very ineffective feedback.

The average amplifier/loudspeaker combination struggled to 4 or 5KHz; the better ones made 7 before the rot set in!

Nowadays, it's the poor systems of the hi-fi breed that merely make 7KHz; the better ones carry right on and, with tweeters, get up to and beyond the limit of audibility.

It was actually the appearance of this kind of loudspeaker which spelt trouble for hi-fi enthusiasts at the tail end of the 78 era. With their specifications, the speakers were irresistible; with the kind of noise and distortion they high-lighted off the records, they were just about unuseable!

That is why, circa 1945-54, we were concerned about sharp cut-off treble filters. Most of these were intended for connection in the voice coil circuit of the loudspeaker, because of the inconvenience of breaking into the amplifier chain proper and the likelihood of inductors in low level circuits picking up hum.

What was true then is still true today; If you want to reproduce 78 r.p.m. records successfully through a wide range system, you will need to cut off the response sharply above certain frequencies - say, 4.5KHz for very poor records and 7KHz for better ones.

If your amplifier system is one of the few with genuinely sharp cut-off facilities, your problem is solved.

If you have a multiple loudspeaker system with half-section filters and / or woofer that actually rolls off above about 5KHz, you need only switch out the tweeter(s).

If you have a multiple cone loudspeaker and/or can't get rid of the treble response otherwise, then we can only suggest that you make up one of the voice coil filters referred to earlier - or two if you insist on using both channels of a stereo system.

The information on these filters is summarised in the accompanying panel. As will be apparent, the cut-off is very steep, allowing the essential musical content of the discs to be retained, while sharply attenuating spurious signal in the way of noise and distortion.

To this point, we have not paid any attention to playback compensation - a matter to which our correspondent drew special attention. The reason simply is that we do not regard it as the main barrier to acceptable reproduction of old records.

As we have been at pains to point out, the main barriers are the frustrating quality of mid-thirty sound to mid-sixty ears, the problems of stylus fit and most important of all, the extended treble response of modern reproducing systems.

In particular, this latter has more to do with the extended response of present day wide range loudspeaker systems than with amplifier design as such. There would be little to be gained, therefore, by resorting to an old amplifier and using it with modern loudspeakers. The more logical approach is to use any suitable amplifier - modern or otherwise - but with a loudspeaker system with treble response limited either by its very nature, or switching out the tweeter or by the insertion of a sharp cut-off filter.

Only then is it worthwhile to worry about compensation curves.

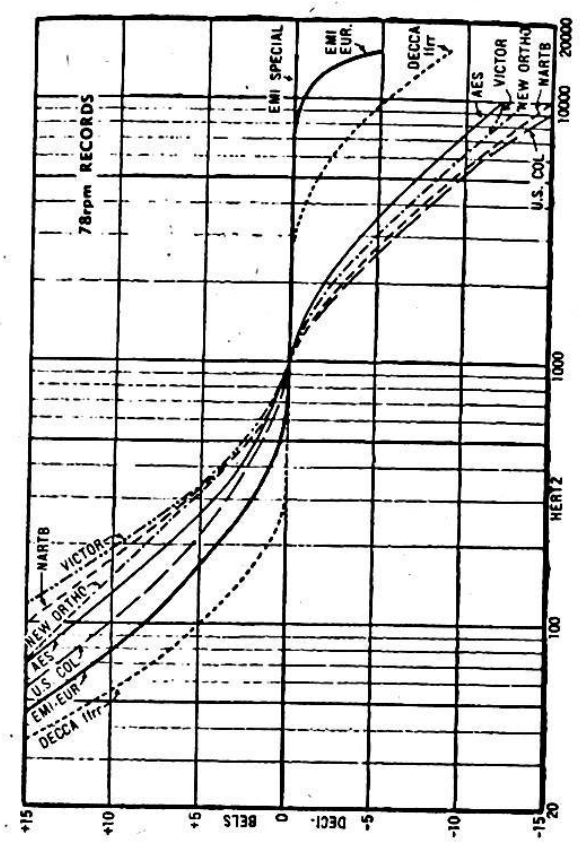
The forlorn nature of our correspondent's hope for a "near enough" curve will be apparent from the accompanying set of curves which show the playback response called for by the electrical recordings which were current at the time when the data were collected. It makes no attempt to cover the hotch-potch of practice which was current during the days of mechanical recording, when engineers were far more concerned with major obstacles than with the finer points of constant velocity and constant amplitude - the stuff of which recording characteristics are made!

In any case, it does not follow that optimum sound will result from optimum compensation - even assuming that one has been able to determine this by examining the pedigree of a particular disc. It may well transpire that the sound can be made most enjoyable by an arbitrary juggling of bass and treble having regard to the amount of rumble and surface noise which happens to be present.

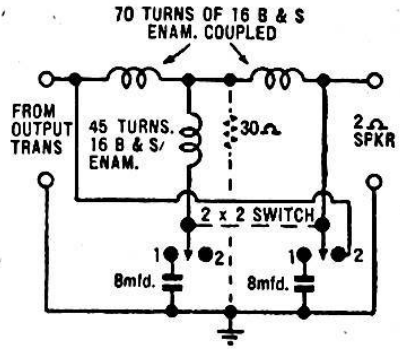
In short, one can make a good case for playing the discs through an ordinary RIAA channel, as used for modern microgrooves, manipulating the bass and treble controls thereafter the most pleasing sound.
The average treble control can easily bridge the difference between the U.S. inspired characteristics (AES, NARTB, etc) while application of treble boost can cancel the pickup de-emphasis to approximate the Decca ffrr or the "flat" EMI characteristics. Remember that the system should have been rendered insensitive, anyway, to frequencies above 6.5 or 8.5KHz.
At the bass end, the bass control should likewise offer a fair range of adjustment to take up the difference between the U.S. inspired curves. Some bass cut may be necessary to get back towards the EMI and Decca characteristics. Ideally, these call for a different contour, with the bass boost commencing much lower down in the register. If your amplifier system has a "78" playback setting, there is every chance that it will give more like this kind of correction but modification will still be called for by the bass control.

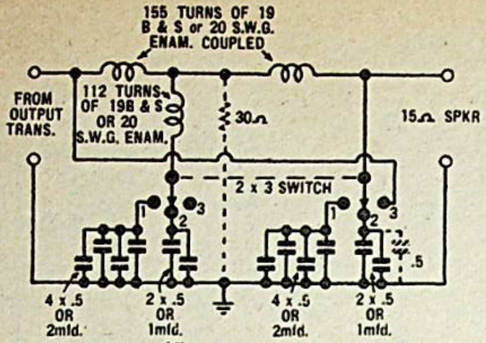
VOICE-COIL TOP-CUT FILTERS

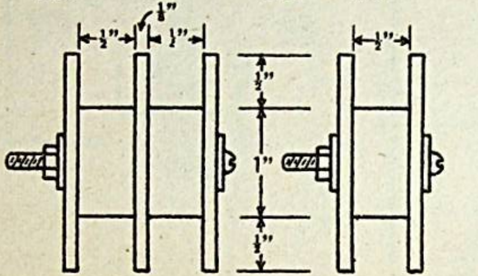
Summarised are the data for top-cut filters for 15.8 and 2-ohm voice coil circuits. In the first two, positions 1 and 2 give a roll-off respectively at 6.5 and 8.5KHz; position 3 is straight through.
The 8.5KHz cut is useful on radio to combat the 10KHz inter-station whistle and chatter. It is also useful to combat mild distortion from old recordings. The 6.5KHz cut will cope with all but bad records.
Layout of the filter is not critical but it is most logically housed in a non-metallic box. The twin coils should be wound in the SAME direction with the outside of one connecting to the inside of the other and forming the common connection. The single coil should be mounted remote from them and at right-angles. The coils can be hand-wound but as near to layer-wound as practicable, with care to achieve the specified number of turns. Bobbins must be non- metallic and must not be mounted flat against metal surfaces. Wire should be of the specified gauge or, at most, not more than one gauge from it.
Capacitors can be to the values marked, or can be made up by connecting smaller units in parallel. They must be paper types (not electrolytic) and should be bridged to see that their capacitance is within - 10 and + 20 per cent of the required value.
A resistor of from 30 to 50 ohms, shown dotted, helps to flatten the filter when operating into a loudspeaker load; increasing the output capacitor by about 25 per cent or to a maximum of 50 per cent will steepen the cut-off if required.
Also, shown is the same filter configuration with constants modified to suit an 8-ohm circuit. The use of a more complex switch economises in capacitors by using them singly or in parallel as required.
A 2-ohm equivalent presents difficulties because of the high value capacitors required but a unit giving a single, compromise roll-off at 7KHz is shown.



Drawn up in the days when we were concerned about compensating 78rpm records, this diagram indicates the variety of playback characteristics which were called for, in theory at least.

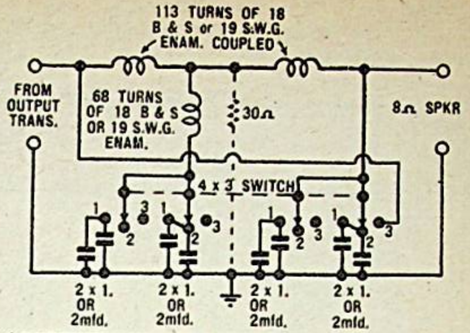


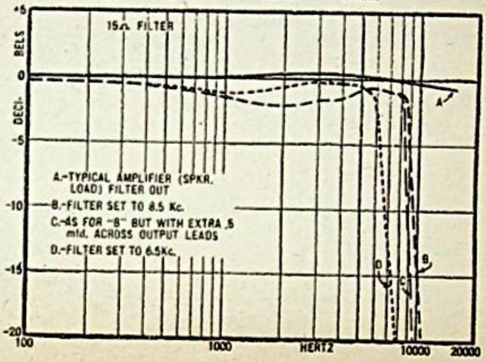




155 TURNS EACH
OF 20 S.W.G. ENAM.

112 TURNS 20 S.W.G.
ENAMEL





STEREO RECORDS: SPREAD OR SEPARATION?

When playing stereo records in the home, what should one expect to hear? A very obvious separation between channels or something not so obvious? What is "three-channel" stereo and "reprocessed" stereo? These and other questions are answered in the following article.

Before answering the questions, it may be appropriate to make some observations about stereo records in general.

In a stereo disc system, two distinct audio signals are recorded in the spiral, V-shaped groove. Without becoming too involved in detail, it is not far from the truth to say that the two signals are impressed on the respective groove walls. The resultant is a rather complex groove, which is modulated both laterally (side to side) and vertically (in depth).

When the groove is traced or replayed by a suitably designed stereo pickup cartridge, two distinct audio output signals are produced, which are substantially a replica of those which were originally applied to the recording head.

The signals from the pickup cartridge are fed into two separate amplifier channels and thence into two separate loudspeakers or groups of loudspeakers in the listening room. Heard in combination, the two sources can impart a sense of direction and dimension to the reproduced sound that is not present when only single-channel reproduction is available.

If the potential benefit of stereo reproduction is to be realised, attention has to be given to the placement of the separate loudspeakers relative to the listening position. This is sometimes regarded as an imposition on domestic furnishing arrangements but, in fact, it is no more so than the need to arrange seating in relation to the television receiver.

If the living-room is to double as an entertainment centre, as distinct from a mere exercise in furnishing, it is logical to arrange the layout with due attention to the video and audio sources. This leads fairly naturally to the concept of having the stereo loudspeakers and the television receiver arranged along one wall, with a suitable proportion of the seating facing it along the opposite wall.

In an ordinary living-room, say 18ft by 12ft, it is appropriate to have the loudspeakers near the ends of one of the 12ft walls, with the seating grouped at the other end of the room. If most of the listening is done by a couple of people only, then can usually sit fairly centrally, with the loudspeakers angled slightly inward to face them. If more people have to be accommodated, it is sometimes found better to angle the loudspeakers more sharply inward, so that people around the sides of the room get more direct radiation from the loudspeaker which is furthest from them.

As far as possible, the area immediately adjacent to the loudspeakers should be uncluttered by adjacent furniture, to allow the wavefront to spread uniformly across the room. It is not a good idea to have a couple of small enclosures tucked down unobtrusively at either end of a sideboard and further obscured by drapes and furnishings. They should be elevated to where – admittedly – they may be seen but, more importantly, to where they will be heard.

The loudspeakers used in the respective channels of a stereo system should be of the same type or, at the very least, should have a similar performance over the middle and high frequency part of the spectrum. If the systems are notably dissimilar, the sound radiated from them will tend to distribute, not just on the basis of the signals fed to them, but to some extent on the degree to which frequency components may happen to coincide with peaks and troughs in their individual response curves.

A second channel supplied by a poorly matched loudspeaker may certainly be better than no second channel at all, but it will prejudice the listener's ultimate assessment of the stereo effect.

Matched or unmatched, however, the stereo effect cannot be created properly in a room by loudspeakers, which are merely three feet apart, at opposite ends of a conventional stereogram cabinet. Heard from an ordinary and convenient listening distance, it would be fairer to designate the sound as augmented mono, rather than true stereo. By and large, the remarks which follow assume that the listener has installed a system with separate, suitably placed loudspeakers, preferably of the wide-range variety.

A matter which is sometimes overlooked is the need to ensure that the loudspeakers operate in phase. This simply means that, when fed with a similar audio signal, the cones should move forward and backward in unison. A full article on the subject of loudspeaker phasing appeared in these columns in the February 1969, issue. (Copies of the issue are available from our Back Dates Department for 50c each.)

Also important in the matter of stereo separation is the ability of the pickup cartridge to derive separate and distinct audio signals from the stereo groove. All practical cartridges suffer to some degree from what is known as "crosstalk" – a tendency to reproduce signals intended for either channel, very weakly in the other. Provided the ratio between the deliberate and incidental signal is 15dB or better (a signal voltage ratio of 5:1 or greater) the stereo effect is not markedly impaired. Most modern cartridges will equal or better this figure over most of the frequency range but, if stereo separation in a system is markedly poor, some suspicion must attach to the cartridge being used.

In this connection, readers' attention is drawn to a record which has been on the market for some time and which is just about ideal for observations of this kind. The first track is a recording of a bouncing ping-pong ball, which should be heard first from the left-hand loudspeaker, then from the right-hand loudspeaker. Later it should seem to come from the area between them. Even in a high-quality system, some cross-talk will be evident to a listener crouched in front of the "silent" channel but, from a normal listening distance, the directional illusion should be complete.

The record in question is issued by Festival under the little "Miracle In Sound", Festival Stereo Sampler SFL 2/1. Other than the bouncing ball, the tracks are all musical excerpts, all pleasant in themselves. It is stocked by most large record retailers or can be obtained readily on order.

Now for some of the questions posed at the beginning of this article – for example: What should one expect to hear from a stereo record?

Over and above the technical challenge which stereo posed to engineers, there seems little doubt that the original and the "purist" intention was to develop a system which would reconstitute as closely as possible the spatial spread of an orchestra, as it would be heard by a listener somewhere near the centre of a concert auditorium. Instrumental sound coming from the listener's left or right in the concert hall, would seem to come from similar directions when the same sound was reproduced in the home. Similarly, sounds which originated nearer the centre of the orchestra, would seem to come from a similar region between the two loudspeakers.

The classic way to record an orchestra with this objective in view is to suspend a twin microphone assembly fairly centrally above the early rows of seats in the auditorium. Each microphone is fairly directional and the two units are angled to cover the respective halves of the orchestra.

Sounds produced in the orchestra are "heard" by each microphone but, because of their different orientation, there is a suble difference in both the amplitude and the phase of the signals which each microphone delivers to the stereo recorder. Intermixed with the direct sound, of course, is a two-microphone version of the reverberant sound from the auditorium.

Experience and expertise play a big part in recording an orchestra for the desired stereo effect but the end result can be extremely good. The orchestra appears to spread right across the end of the listening room, with just as much sound seeming to come from between the loudspeakers as from the loudspeakers themselves. In fact, if the loudspeakers were not so plainly visible, their position would not be at all obvious.

This kind of result is normally sought as a matter of course, in stereo recordings of ranking, classical orchestras, irrespective of the country of origin. It is therefore the kind of sound which you can reasonably expect to hear from any stereo recording of a symphony style orchestra, made by a company which is active and experienced in the classical field . . . smooth sound, evenly distributed across the whole listening area.

Classical records which do not fulfil this expectation come mostly from recording companies which are mainly active in the popular field and which are likely to apply other than "purist" thinking to the production of any records, popular or classical.

If your conviction is that you have never heard smooth, evenly spread sound from a classical recording released by a company experienced in this field, then something is very wrong somewhere. You are certainly missing out on what a great many other people enjoy.

Possibilities to consider are:

- The loudspeakers are incorrectly phased. See the earlier reference to this subject.

- Separation in the cartridge is inadequate.

- If the spread seems quite inadequate, your loudspeakers may be too close together. Alternatively, they may be so cluttered by furnishings that listeners in the room only ever hear one of them properly at a time.

- If the sound always appears to come from the two separate loudspeakers and never from the area in between, the loudspeakers are possibly too far apart. The trouble can be compounded by a room which is over-heavy with carpets, curtains and soft furnishings.

- It is just possible that there is nothing wrong at all with the system acoustically and that your eyes (which say that there are only two sound sources) are over-ruling your ears.

On this last point, a few enthusiasts have insisted in installing an actual centre amplifier channel and loudspeaker, fed with a signal artifically compounded from the other two. While the scheme is technically quite legitimate, the majority opinion is that a physical centre channel should not, in fact, be necessary for appropriately recorded discs.

If, for one reason or another, your record library contains more than its fair share of formal music recorded with inappropriate separation, a centre channel might be justified, but scarcely otherwise.

As distinct from what has been referred to as the "purist" method of recording stereo, an alternative approach has been to position a number of microphones in front of, or within an orchestra, directing their outputs through a multi-channel control console on to separate tracks of a multi-channel tape recorder.

As a separate and subsequent operation, the content of the various tracks is mixed onto two tracks for transfer to disc in such a way as to achieve the balance, spread and emphasis that is judged to be appropriate by those musically responsible for the final production.

Many and varied have been the arguments for and against alternative methods of recording stereo but, in terms of the end result, an evenly spread sound can be obtained from a stereo microphone assembly, from multiple microphones, or from a combination of both. If the producer so desires, separate microphones can be used to provide the facility – the desirability of which is often debated – of giving emphasis to soloists or sections of a performing group.

However, the story of stereo does not end at this point. Equipped with multiple microphones and multiple channels, many arrangers and producers have elected to exploit the potential separation of stereo channels to provide the listener with two virtually distinct sounds, which are projected almost independently into the listening room.

A variety of possibilities is open to the producer. A vocalist may be projected through one channel, their accompaniment through the other; the vocalist may dominate one channel, with the orchestra spread out in traditional fashion; instrumental groups can be sub-divided, choirs can be sub-divided and so on. It is quite commonplace these days for producers, seeking a high degree of isolation, to use acoustic screens, separate studios or over-recording techniques.

This contrived separation between the two channels is commonly referred to as "gimmick" or "ping-pong" stereo – a term which has probably been derived from the kind of recording mentioned earlier, where ping-pong sounds are used to distinguish the channels for test purposes.

As might be expected, the distinction between channels is more obvious from a record which emphasises separation that from one intended to produce a sense of spread – particularly when replayed on a single-unit stereogram with loudspeakers only two or three feet apart. In fact, people who use such equipment, and who know no better, are likely to reach the mistaken conclusion that certain popular style records achieve recognisable stereo but that the rest are very little different from straight mono!

As an extension of the highly separated "ping-pong" type of stereo, some arrangers and producers have come up with a sound which reviewers are fond of describing as "three-channel" stereo.

This involves recording distinct sections of the accompaniment on each of the two stereo tracks as for ping-pong stereo. In addition, however a vocalist or featured artist is recorded on a separate microphone and an equal and in-phase signal from it is mixed into both channels.

When reproduced, through a properly set-up system, the equal in-phase, common signal creates a sound that appears to come from the space between the two loudspeakers, exactly as happens when a mono record is played through a balanced stereo system. At the same time, the two channels contribute their own distinct sounds and the total effect is that of three channels – left, right and centre.

It would appear that many listeners do not appreciate the differing intentions and methods by which stereo records are produced and one often hears critical comments about records in which the stereo sounds are not separate or conversely about records in which they are – as if one presentation or the other was completely right and the alternative was completely wrong.

Nothing of the kind.

The simple facts are:

- In many stereo records, the aim is to re-create an even spread of sound across the source area. The producers seek to avoid a hole-in-the-middle and dramatic separation as effects which would be inappropriate to particular types of music. Discs of this type will logically include classical works, brass band performances, most large choirs, and large orchestras in more formal performances of middle-of-the-road music. In no sense does this exclude other material, of course.

- Many other records are deliberately intended to exploit stereo separation, emphasising two or an apparent three channels. This approach is most commonly found in popular vocal and instrumental releases where performance, arrangement and manipulation of the stereo facility are all used to gain maximum sonic impact. Such records are not supposed to produce an even spread of sound and it is debatable whether there is any point in trying to achieve this effect from them.

What then should a listener to a stereo record expect to hear?

Perhaps the best answer is simply: "What the producer expected them to"!

If the reproducing system is properly set up, a record intended to give even spread should produce that kind of spread. Similarly, a record intended to give a two-channel or three-channel effect should sound that way – exactly.

What if the listener believes that an even spread of sound isn't dramatic enough?

His only resource will be to buy records from labels which specialise in the other approach. There will be no dearth of titles in the popular field but his choice of more formal music will be limited.

What if the listener dislikes "gimmicked" separation?

He should buy the records that his opposite number rejects. If his supply of percussion and pop turns out to be too limited by this course, his only resource will be to provide an extra channel fed with a sample of both signals, in an effort to correct the misdeeds of erring producers!

What is – or was – "stereo action"?

This was a gimmick exploited by RCA in particular, although there is no special reason why other companies should not have produced similar effects. By using multiple microphones and manipulating the appropriate controls, it is not difficult to switch a performer, an instrument or a group of instruments from one channel to the other. Alternatively, by gradual cross-fading, the sound source can seem to slide across stage, just as if the vocalist, the piano or the percussion group was mounted on a silent trolley. It was an intriguing idea but not one that contributed much to listener appeal.

What is "reprocessed stereo"?

Almost invariably it signifies an original (and usually old) mono recording which has been re-recorded through a stereo amplifier/cutter system, with some effort made to contrive a difference between the signals fed to the respective channels.

Most noteworthy examples have been a few records which have been painstakingly reprocessed using frequency filters, reverberation circuits and manipulation of the console controls to give an illusion that certain instrument groups were located predominantly to right, left or centre.

However, this painstaking approach is too costly for anything but prestige historical performances, for which a high re-sale rate can be anticipated. Most reprocessed or simulated or reconstituted stereo records are made simply by feeding the signal through filters so that the high frequencies predominate on the left – where the lead strings are traditionally located – and the lower frequencies predominate on the right. Sometimes, artifical reverberation is added in phase to both channels to combat the acoustic "dryness" of old recordings and to give the illusion of distance and depth behind the frequency – separated channels.

Despite their artificiality, reprocessed stereo records can be quite acceptable as mementos of historical performances. In fact, being re-recorded from the original "mother" or from a non-grainy pressing, and with the benefit of discrete compensation, frequency division and reverberation, they can sound a good deal better than commercial shellac pressings of the day,

Can stereo be accurately reconstituted from an original mono recording?

Many would-be inventors have claimed that it can be methods which have ranged all the way from playing two pickups in the one groove to disposing multiple loudspeakers around a room.

Such methods can produce various effects with various types of sound, which may or may not be considered worthwhile; this much is not denied.

However, no ordered method can recreate information about random events, which were not recorded at the time. Simulated stereo may have a little more ambience than the original mono recording and sound a little less confined but it cannot have the spread or the definition of an original stereo recording.

And, finally, how should a mono recording sound on a stereo system?

In a highly symmetrical stereo system, heard from a position equidistant from the loudspeakers, the sound should appear to come from a point midway between them. If there is a TV set in the mid-position, voices in particular will appear to come from it, as if it contained the only operating loudspeaker. The more starting the effect the better is the balance of the system.

With music, particularly complex music, the illusion will be less strong, partly because the widely ranging frequencies may discover slight differences in the response of the individual sections of the cartridge and the individual loudspeakers. And, within the room itself, differing reflection paths will tend to strengthen sound from one loudspeaker or the other over individual portions of the range. If anything, this slight random dispersion of the sound tends to add acoustic interest and most people prefer to listen to mono records in the stereo mode, rather than to disable one channel.

STEREO BALANCE IN DEBATE

The matter of stereo balance continues to be the subject of comment, both related and unrelated to what has been said before Some good purpose may be served, therefore, by acknowledging other opinions and additing relevant remarks.

To start things off, a reader makes some observations about mono sound systems which are quite pertinent to a idscussion of stereo balance – as you will probably come to agree.

He refers to his experience of public address systems installed in public halls, where a central microphone and a single amplifier feed two loudspeaker systems, placed symmetrically on either side of the stage. He points out that, when properly phased and balanced, the reinforced sound appears to a person in the centre of the hall, to come from the vicinity of the microphone, neither loudspeaker being heard separately.

To use his own words:

"The effect of no apparent sound source is only present when the sound intensity is properly balanced. In one case of a public hall, it was possible to walk up the centre of the hall for about 50 feet with the sound appearing to come from inside one's own head.

"Stepping to right or left localised the sound source as the loudspeaker on that particular side of the hall.".

Having, in recent months, had the opportunity of experimenting personally with just such a sound system, I agree in principle with what A.M. says.

In the sound system to which I refer, speech reinforcement is delivered into an auditorium from two symmetrically placed but completely camouflaged sound columns. Using a good quality microphone and amplifier, and with the level set for comfortable listening, for much of the time the audience is completely unaware that the speech is being reinforced.

Down the approximate centre-line of the auditorium, the apparent sound source is at the microphone – a position which it could not occupy physically, because of acoustic feedback problems.

Even at some distance away from the centre-line, the apparent sound source stays fairly central. largely, I feel, because there is no visual evidence to prompt or support any aural conviction that it might be otherwise.

Observations along these lines will doubtless have been made by many of our readers and, while they involve a purely single channel or mono situation, they do indicate that signals from two misplaced sources, properly balanced and phased, can produce the effect of sound coming from a point midway between them.

A.M. sums it up thus:

"It was not until I heard stereo that I fully appreciated the effect, because true stereo involves this same phenomenon, deliberately invoked."

Based on this line of reasoning, A.M. goes on to say that he has always been greatly puzzled by references to "the hole in the middle" since, with proper balance and phasing, that is the very place from which ALL centrally located sound should appear to come.

In his own case, he says there are three positions in his room from which he conventionally listens, according to what he happens to be doing at the time. The interesting thing is that, although the positions are six to eight feet apart, he can always obtain good balance by properly adjusting the relative loudness from the two loudspeakers.

Basically, I go along with these observations, having had no reason personally to complain about "hole in the middle" effects from proper stereo recordings. (I will qualify my use of the word "proper" a little later.)

Plenty have complained about the "hole in the middle" effect, of course, and I am inclined to blame three factors for most of these complaints, singly or in combination:

VISUAL: To have two bulky loudspeaker enclosures in full view, when listening to stereo, is a constant visual reminder that the sound sources are there and there, irrespective of what the ears would indicate. The more the loudspeakers can be merged into the general decor, the less will eyes contradict ears. On this basis the "wall of sound" idea, exploited in hi-fi showrooms and audiophile dream-homes, makes good sense. Happily enough, it can also tidy things up from the point of view of the "little woman", if a full-width speaker fret can be combined with bookshelves above and broken up internally to provide space for equipment and record storage.

WRONG PLACEMENT: Stereo speakers can be too far apart relative to the listening distance. If each ear becomes predominantly conscious of the adjacent loudspeaker, the sound may not be merged as well as it should. By and large, the angle subtended by the speakers at the listening position should not be more than 90 degrees and preferably nearer 60 degrees.

UNBALANCE: I am not referring here so much to loudness balance, since this is fairly obvious and, in any case, will tend to shift the apparent source of wound one way or the other without creating a "hole in the middle". I refer to balance in terms of frequency response, both of the loudspeaker units and that imposed by their immediate external acoustic environment. If serious unbalance is present, would will appear to come partly from one speaker, partly from the other, depending on which happens to have the advantage in terms of efficiency, from one musical frequency to the next. Thus, sound which should balance and occupy apparent centre, may appear to be seriously dispersed.

Also commenting on the "hole in the middle" effect, or rather the lack of it, a reader H.F. says that he has a stereo system in a 24ft x 12ft living room, with speakers against one of the shorter walls, on either side of a fireplace.

Playing a mono record, he says he can move the apparent source of sound at will from one speaker, across the intervening space to the other.

With the channels balanced, sound from a mono record appears so definitely to come from the fireplace that he has has the experience of a visitor examining the area round the heater in search of a third and hidden loudspeaker!

He says:

"It would appear that the trouble experienced by your correspondents would at least partly disappear, if not so much concentrated upon".

In my own case, it happens that an electronic organ consile stands between the two stereo enclosures. Much of the time, with a centrally placed soloist, and particularly with a mono record, it is difficult to persuade oneself that the organ speakers are not in fact, reproducing the sound.

Reader H.F. also points out the necessity of having the speakers in phase as a matter which has been stressed many times in these pages and elsewhere.

He makes the further point that volume controls should be matched in terms of resistance per angle of rotation, so that the balance will not change significantly with variations in volume control setting.

Reader R.C. takes up the same point in connection with stereo balance. He says that he has noticed that volume control settings alter the balance, in some cases, to a quite noticeable extent.

"I consider that the main cause of this trouble is the ganged volume control and treble control potentiometers.

"I have found that the word 'ganged' seems to imply merely that a common shaft is used to operate the two potentiometers, rather than to mean that the values change at an equal rate with different settings. This unequal rate of change in values can offset the balance to a quite noticeable extent."

One can hardly deny the truth of this statement, as it is fairly self-evident. What is not so clear is the degree to which the effect is evident in practice.

As I think I have pointed out in the past, the average audiophile tends to play most of his records, most of the time, at a certain volume, at which volume setting the balance control would logically be adjusted. This done, the important matter is not any discrepancy in the volume control elements, checked over the whole range, but the variation which occurs from proper balance over the smaller sector of the controls which are used in practice.

It might easily be that unbalance in the treble response is the more significant factor, since there is no provision in the average amplifier to correct discrepancy between channels in this respect. Should the discrepancy be marked, it could cause some components of a "centre" signal to move to the side having higher treble output. In short much the same effect could be evident as the signal "dispersion" on a frequency basis, mentioned in connection with dissimilar loudspeakers or loudspeaker environments.

Considering the possibility of frequency unbalance in the pickup as well, this could add up to a good case for checking treble response of the two channels, preferably from frequency disc to loudspeaker voice coil.

Some have suggested that all such objections could be overcome by having tandem rather than ganged volume and tone controls and, while the modification would be purely a mechanical one, I still have the same reservations as expressed some months ago. I just don't think that it would be practical to maintain the balance of the controls accurately while using them on ordinary stereo program signal.

I would far sooner rely on a ganged system, balanced, if need be, over the range where it is most likely to be used.

One recent correspondent does support the idea of tandem rather than ganged controls, but for a different and rather interesting reason. Writing M.B., says:

"I have noticed that you do not agree with separate bass and treble controls in each channel of a stereo system. I quite agree with you.

"However, I would like to tell you of a way in which I improved the sound of mono. discs through a stereo system, using separate bass and treble controls.

It adds realism by the fact that you can have the drums or bass section of the mono. disc coming from the left speaker system and the higher notes from the right speaker, or vice versa. This can not be done when you have ganged bass and treble controls, as in most systems.

"Of course, when you are playing stereo records, you have to make sure that the bass and treble controls are kept strictly in step".

Fundamentally, there is nothing new about this idea, since it was the basis on which hi-fi enthusiasts tried to simulate stereo effects long before stereo signal sources became available.

However, the idea may not have occurred to those whose interest dates back not quite so far.

But to round things off, let me add a few remarks about stereo discs as they are in practice – not as we sometimes imagine them all to be. Hence my earlier reference to "proper" stereo.

The original idea of stereo reproduction was to simulate, as accurately as possible, the dispersion of sound sources as heard by a listener, facing an orchestra, for example. Various microphone and mixing techniques have been used to achieve this end, and it can be achieved to a high degree of satisfaction.

Stereo of this nature and for this purpose probably predominated in the classical recording field.

In the less formal fields, music is recorded in this fashion on only a proportion of all discs and I wouldn't even be prepared to suggest that it would be a major proportion.

In a very large number of discs the two channels are used to separate one part of the music from the other – soloist from accompaniment, and so on. To emphasise separation, recording engineers at times physically separate players, leaving them just enough "communication" to maintain cohesion in the performance.

Sometimes the motive is to produce a sonic duet; sometimes merely to produce a disc which will seem to have reasonable separation on the many "stereograms" with speakers much too close together.

With all this "built-in" separation, it is clearly futile to worry about the "hole in the middle".

Of course there's a hole in the middle. That's the way the recordist or the producer made it!

Equally, I have referred to the occasional disc where one has the impression that the recordist has sought for three precise sound sources – one at each speaker and one in the centre, with as little in between as possible.

So, before you generate too many complexes about "holes" or no holes, make due allowance for the original recordings.

Play a vocal mono disc with the system set for stereo and see whether the artist can be made to sing for you from the space between the speakers.

If you're worried about eyes contradicting ears, put a dummy speaker in the middle and listen. If you can make the sound seem to come from the dummy speaker, there can't be too much wrong with the balance of your system.

If you can now find a stereo record which gives the proper sense of spread it will at least demonstrate that the system is capable of "purist" stereo.

If some of your other records leave a hole in the middle, don't automatically blame the system. In the "pop" field particularly, there's a good chance that the records were made that way.

Does this mean that you've been "taken?"

No, certainly not! It simply means that some recordists and producers think differently from you. They prefer to use a two channel system to give them two channels rather than spread in the "purist" sense of the term.

And having said this, it's not much of a step to see the questionable value of stereo output meters. About the only time they can give a completely valid reading is when playing a mono record through the "stereo" setting. Once a stereo record is put on the turntable, the output of the two channels must be different and the more the record is gimmicked toward two separated channels, the greater will that difference be.

So we have the rather anomalous position of a stereo balance indicator which is of most value when not used for stereo!

STEREO PICKUPS, VERTICAL TRACKING

During the last year or so, there has been a good deal of discussion, in technical circles, about the vertical tracking angle of pickups, with frequent mention of the figure "15-degrees". This article explains what it is all about, in practical terms.

Prior to all the recent discussion of the subject, the term "tracking", applied to a gramo pickup, was assumed to refer to the alignment of the head assembly with a tangent to the radial groove at the point of stylus contact (Figure 1). In short, LATERAL tracking was understood, although the adjective was implied far more frequently than it was used.

For precise lateral tracking, it was assumed that the pivot axis of the moving assembly should be above the tangent and in line with it, so that the natural side-to-side movement of the stylus tip, viewed from above, would be at right angles to the groove direction at the point of contact.

For the meticulous, the concept was somewhat complicated by the angle at which the stylus assembly approached the record surface. Cantilever styli called for a somewhat different concept again, although it was usually conceded as sufficient if the cantilever arm lined up, with a tangent drawn through the point of stylus contact.

Any digression from the ideal was – and is – referred to as LATERAL tracking error.

In practice, it is not possible to obtain perfect lateral tracking with the usual rigid pickup arm, pivoted at one end and describing a radial track across the record surface. However, by offsetting the head axis with respect to the arm axis and by carefully predetermining the offset angle, the length of arm and the distance between the pivot and turntable spindle, the tracking error can usually be kept to within two or three degrees plus and minus.

This is normally regarded as satisfactory.

The angle which the stylus made to the record surface, viewed in evaluation, naturally received its share of attention at various times and for various reasons.

As viewed end-on – from the front of the pickup – and is normally accepted that the stylus assembly should make a right angle with the surface of the record.

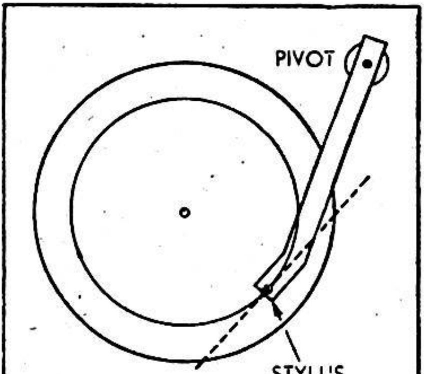


Fig. 1

Lateral tracking angle has been the subject of close attention from the days of early acoustic phonographs. It is perhaps more important than ever with stereo systems.

Viewed from the side, however, the optimum angle is not self-evident.

In the days of "needles", whether thorn or steel, it was accepted that the stylus should have a trailing angle, so that it pointed obliquely downwards and away from the base of the pickup. It was reckoned that wear on record and stylus alike would be less than if the stylus were vertical or, worse still, given a negative angle such that it would tend to "dig in".

While the majority of pickup designers thus settled for a trailing angle of somewhere around 20 degrees from the vertical, there were some notable exceptions. Those with long memories may recall, in particular, the "long play" needles which made their appearance, with a kink in the middle to increase their trailing angle – along with their mass, one might mention, and effective "tip" dimensions!

At the other extreme were the high fidelity pickups of the immediate post-war period, with the emphasis on reduced stylus mass and dimensions and, reduced – or near zero – trailing angle.

With the later and general adoption of spherically ground sapphire and diamond styli, ideas about "digging in" and the consequent apparent need for a positive trailing angle had to be revised or dropped.

If the tip of a stylus is considered as spherical in form, wedged between the sides of a V-groove, the groove encounters substantially the same configuration, whether the sphere is being held from directly above, or from a position which would represent a trailing or a leading angle.

Relieved of the more obvious crudities of dragging a steel needle along several hundred feet of abrasive groove, designers were able to pay more attention to the finer points of record wear, stylus wear, noise reduction and groove tracing – including "pinch effect", the tendency for a stylus to ride over curves rather than around them.

Not surprisingly, therefore, the last generation of mono pickups showed little regard for earlier conventions of stylus angle.

With the arrival of stereo, a new and vital consideration arose.

In the stereo system, information was contained, not only in lateral deflection of the stylus but in vertical as well.

But what does vertical mean in this context? Truly vertical or just nearly vertical?

The "obvious" assumption is a cutting head in which the stylus assembly is so suspended that it moves directly up and down when fed with "vertical" signal information. In fact, a large section of the European recording industry worked originally on this assumption, while the American industry by and large, followed a different line.

In typical U.S. stereo cutters, the stylus assembly is attached to a support arm, with an actual or virtual pivot point somewhere inside the body of the cutter and therefore somewhere above the record surface. As illustrated in Figure 2, this causes the cutting stylus to travel through a short arc, of which no segment can possibly be vertical.

The popular Westrex cutter exhibited a nominal 23 degrees of inclination but American engineers found, within their own industry, examples of inclination all the way from 10 to 40 degrees.

When the position of a stylus, so anchored, is "vertically" modulated, the slight horizontal component in the movement, due to the tilted locus, either adds to or subtracts from the relative rotational speed of the blank being inscribed. As the stylus lifts, it moves with the blank and reduces the relative groove speed; as the stylus lowers, it increases the relative groove speed. This imparts a slight tilt to the waveforms.

Initially, the tilt effect was dismissed by most as academic and of little practical significance to the end result, as heard. The physical details of cutters, the attitude of the stylus and the inclination of the moving system relative to the record surface were thus determined by considerations other than the vertical recording angle.

STANDARDS PROBLEM

As a result, practices became established in various centres which were later defended as "standards", when the matter came up for critical examination.

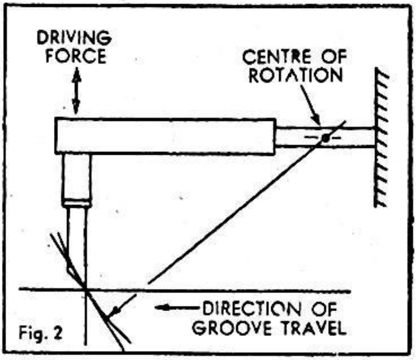


Fig. 2

When a cutting stylus is supported as shown, vertical drive causes it to describe an arc which is inclined obliquely to the record surface.

Much the same situation developed with pickups. Manufacturers were pre-occupied with finding solutions to the urgent and obvious problems of compliance, dynamic mass, frequency response, channel separation and "obvious" sources of distortion. The solutions were sought, without overmuch attention to the exact inclination of the moving system, apart from its relation to other mechanical requirements.

However, as the more obvious problems were disposed of, attention did gradually focus on the question of inclination of the moving system and with it came free use of the term "VERTICAL tracking angle".

Designers began to realise that it was inconsistent to take pains with lateral tracking, and close the eyes to virtually the same problem in the vertical plane.

It was assumed initially that the proper solution would be the obvious one – simply to ensure that cutters and pickups all conformed to the one standard angle of inclination between the locus of "vertical" movement and the record surface – or, if you like, the true vertical. In short, it was assumed that the idea simply involved making pickups (Figure 3) correspond with cutters (Figure 2).

But here certain difficulties arose:

(1) Cutter and pickup manufacturers, influenced by "local" factors, had firm and divergent opinions as to what the optimum or standard vertical tracking angle should be.

(2) Distortion tests aimed at establishing criteria seemed confused and inconclusive.

(3) There was a body of opinion which tended to regard the problem as overstated.

During 1962-3, a series of engineering papers before learned societies and articles in audio journals ventilated the ideas of individual recording groups and substituted important facts and figures for earlier and rather vague speculation.

These papers, by and large, removed any doubts about the desirability of standardising on vertical recording and tracking angles as a matter of broad principle and as a contribution to ultimate fidelity.

At the same time, many engineers expressed doubts about the orders of distortion predicted and whether they had actually been produced in practice.

Some maintained that the vertical component represented the "difference" signal in a 45/45 stereo system and that this signal was of minority magnitude except in records where the two lots of track information were highly dis-similar.

Again, because recording engineers appraised their own work on a listening basis, they tended to avoid practices and recording levels which seemed to produce distortion – irrespective of the precise reasons for such distortion.

Much argument followed, along these lines.

However, it did become quite clear that there were hidden factors which operated against the aforementioned "obvious" solution of building cutters and pickups to the same standard physical inclination. It was these hidden factors which, earlier, had so confused efforts to resolve the matter.

First and foremost, it was discovered quite by accident that the lacquer did not retain the inclination of the vertical pattern supposedly cut into it.

As the stylus passes through the lacquer, it removes a thin ribbon of material, to be sure, but the tip of the stylus also compresses and pushes ahead of it the material forming the V-bottom of the groove. Once the stylus has passed, the slightly elastic material springs back away from the stylus, REDUCING the apparent angle of inclination.

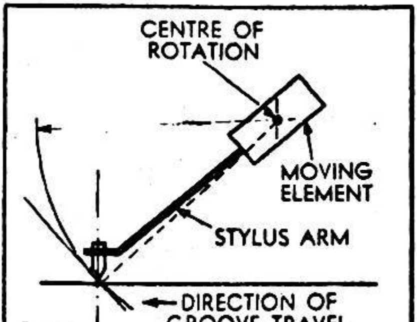


Fig. 3

For optimum vertical tracking, a line drawn from stylus tip to the centre of rotation of the moving element should correspond to the angle of tilt in the recording. This will NOT be the same as the geometric inclination of the cutter system.

By recording square waves as "vertical" information, to produce triangular-shaped groove modulation, engineers found they were able to examine and measure the patterns microscopically and, from them, deduce the geometry of the groove.

They discovered, to their amazement, that lacquer spring-back modified the apparent inclination of the system by more than 20 degrees in a typical case. Thus the Westrex cutter, with a seemingly liberal inclination of 23 degrees, was found to produce a recording with an actual slant of only 2.5 degrees.

And European recordings, made with vertical cutters, had a likely backward slant of 20 degrees or more. The disparity between European recordings, with their inadvertent backward slant, and typical cantilever pickups with roughly as much forward slant was thus apparent. The vertical tracking error was of the order of 40 degrees!

A second order but complicating factor was the tendency of stylus assemblies to bend under pressure. Recording styli tend to bend backward as they cut the groove, while replay cantilever styli are forced upward – relative to the cartridge – by the normal tracking weight. The geometry of both systems is therefore likely to differ under dynamic and static conditions.

To cut a long story short, recording engineers have worked out ways and means of compensating for these hidden factors – changing the design of cutters, modifying their mounting position on the cutting lathe and fitting cutters with styli having appropriately angled faces.

By such measures, there appears to have emerged a majority move to cut grooves which have an effective inclination for "vertical" information displaced from the true vertical by 15 degrees. The figure has R.I.A.A. support.

Pick-up manufacturers who wish to conform to this emerging standard must therefore design their pick-ups so that a line drawn from the stylus tip to the effective pivottal point of the moving system would be inclined at 15 degrees from the record surface.

Putting it another way, a tangent to the tiny arc described by the stylus tip under vertical modulation should make an angle of 15 degrees with the true vertical (Figure 3).

Interpretation of this requirement has led to certain individual misunderstandings and even published mis-statements.

One is that the "shank" of the jewelled stylus, or the tiny flat to which it is attached at the end of the cantilever, should form the requisite 15-degree angles. This has led to suggestions that existing cantilever tips merely need to be bent as necessary to conform to the standard. This is quite wrong because such a modification has only a very minor effect on the relative positions of the stylus tip and the pivottal axis. Tilting the stylus merely presents another portion of the spherical tip to the oncoming groove!

Another is that the cantilever arm should make a 15-degree angle with the surface of the record. This is not necessarily so. At the stylus end the cantilever may or may not line up with the stylus tip; in most cases it will be above it. At the other end, the cantilever may or may not point directly toward the true pivottal point. The angle of the cantilever arm is therefore inconclusive.

Because this is so, there is considerable doubt about advice to tilt existing heads inside the head shell to achieve a particular angle of the cantilever arm. By so doing, one could inadvertently add to the vertical tracking error rather than correct it – quite apart from the danger of "busting" something!

What should be done then about the vertical tracking angle of pick-ups and cartridges currently in use?

The writer's advice is simply . . . nothing in a hurry!

In the first place, many pick-ups in current use may be found to have an effective vertical tracking angle not too far removed from 15 degrees. The information may be available if you look hard enough for it, or it may become available.

Again, if you have a cartridge which is performing satisfactorily, despite a suspect tracking angle, the fact that you've found out about it won't alter its performance. There is no guarantee that you can do much about it anyway!

When your present cartridge seems to have outlived its usefulness and the time has come to buy a new one, that will be the time to really start worrying about vertical tracking angle. My advice would then be to prefer the 15-degree cartridge – all other things being equal.

By that time, a large proportion of new records will be standardised.

But, of course, the word "new" contains the rub. It will be small comfort to readers who have built up a collection of European records which have, more or less inadvertently, been recorded with a backward vertical slant. Standard or no standard, they'll remain the odd men out.

It has been suggested that, to play these records to best advantage, the tone arm should be lengthened and the cartridge turned back to front, virtually necessitating a second pickup. But one would need to be keen indeed to go to such lengths!

AN ECONOMICAL QUADRAPHONIC ADAPTOR

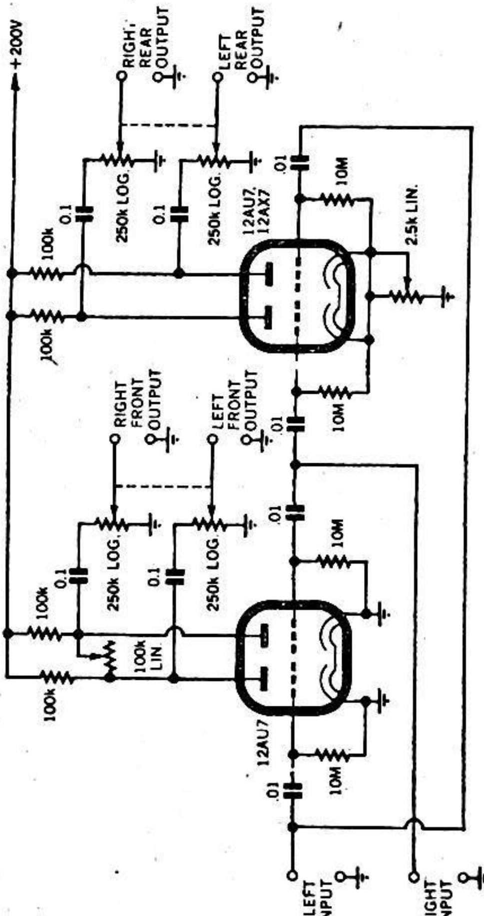
At present, decoders suitable for matrixed four channel recordings tend to be rather expensive. Here is a circuit that can be built up for a very modest outlay. It will "de-code" most types of quadraphonic record such as EV, Project 3, Dyna, etc, as well as synthesising four channel sound from a normal stereophonic disc. Theoretically, it will not provide optimum rear channel localisation with the C.B.S. "SQ" disc but in practice the audible difference is only slight. In common with all four channel systems, an additional stereo amplifier plus two rear speakers are required.

Valves used for the unit are a pair of 12AU7 twin triodes. If the volume is found to be low in the rear channels, a 12AX7 should be substituted for the second valve.

Correct setting of the 100K and 2.5K potentiometers is important. These controls determine the front-to-rear and left-to-right separation. With the 100K potentiometer at maximum resistance and the 2.5K at minimum, the front-left and rear-left speakers receive identical signals, as do the front-right and rear-right. In other words, there is no back-to-front separation.

With the 100K potentiometer at minimum resistance and the 2.5K well advanced towards maximum resistance, we have maximum back-to-front separation but no separation between left and right front, or left and right rear.

Compromise settings must be found that enable slightly different signals to be heard in each channel, to provide a quadraphonic effect. To accomplish this, the following procedure is suggested. Select a stereo recording that exhibits a wide separation and play it through the system with rear channels turned off. While listening from a point equidistant from the front speakers, adjust the 100K potentiometer to the lowest value at which separation is just noticeable. There should still be a sense of left-right location but not extending across the full distance between the speakers.



Next, turn up the rear channels and silence the front. Adjust the 2.5K potentiometer for barely noticeable separation. Increasing the resistance of this control too much should result in an out of phase mono sound! Now adjust the ganged volume controls for proper level in all four channels and the decoder is ready for operation