HI-FI AND AUDIO

MODERN DESIGNS FOR THE AMATEUR CONSTRUCTOR

EDITED BY
A. T. COLLINS

NEW

A PRACTICAL WIRELESS **KEY BOOK** 3/6

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AND
AUDIO

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Contents

High Sensitivity Amplifier

THE high sensitivity of this amplifier, together with its lack of distortion and low noise level, makes it suitable for use where a low input (such as is obtained from certain types of record player pick-up units, and the majority of tape playback heads) is available. The output of the amplifier is more than adequate for domestic purposes.

The Circuit

A novel feature of the circuit, the complete diagram of which is shown in Fig. 1, is the provision of both high and low input impedance facilities.

In the normal way, a low impedance supply is matched to the naturally high input impedance of the valve amplifier by means of a matching transformer. This method has drawbacks, in that the transformer, unless it is very carefully screened, both electrically and magnetically, is a source of hum. In this circuit, therefore, a transistor is used to replace the matching transformer. It is connected in such a way that, while its input impedance is low, its output impedance is sufficiently high to match the valve amplifier. This circuit has the added advantage that the transistor provides a considerable amount of gain.

The remainder of the circuit is fairly conventional. Three

amplifying valves are used. The first is an EF86 working as a voltage amplifier. This valve is particularly suitable in this position owing to its freedom from microphonic effects. This is resistance-capacity coupled to the following stage, a 6BR7—the 6BR7 is a B9A-based version of the better known octal-based 6J7, which could equally well be used in
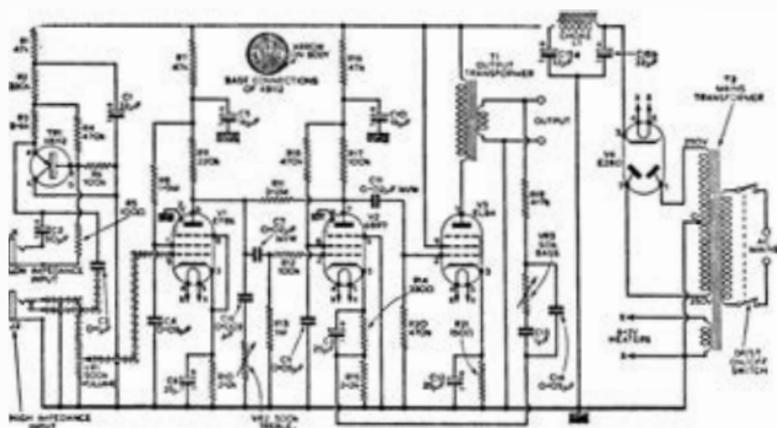

Fig. I. The complete amplifier circuit.

its place. The output stage comprises a single EL84, which is quite capable of giving up to 4·5W output.

Feedback

Volume is controlled in the normal manner, by attenuating the input to the EF86 as required. A simple, but none the less effective, top-cut treble control is taken from the anode of this valve.

Negative feedback is used within the amplifier, both to reduce distortion, and to stabilise the operation of the unit. The main loop is taken from the secondary of the output transformer to the cathode of the 6BR7. This circuit is frequency selective, and by using a variable resistor in the loop as shown, control may be exercised over the bass

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response of the amplifier. This serves as a very efficient bass control circuit. A second negative feedback loop is between the anodes of the EF86 and the 6BR7, via R11 a 2·2MΩ resistor.

Power Supplies

It is very important when designing equipment of this kind, to ensure that adequate smoothing circuits are provided. This is particularly true when, as in the case of this amplifier, the signals handled by the earlier stages are of a low magnitude. To this end it has been found necessary to decouple each stage using fairly high value electrolytic capacitors.

The power unit itself is conventional. An EZ80 rectifier was used in the prototype, but almost any valve, preferably having an indirectly heated cathode, could be used. The mains transformer should be capable of supplying 250-0-250V at 65-70mA, and 6·3V at 1·5A for the valve heaters. If a rectifier having a 5V heater is used, then provision must be made for this when obtaining the mains transformer.

The transistor obtains its power from the smoothed h.t. line, thus eliminating the need for any form of battery.

Components

The transistor used in this circuit is the XB112, and it is recommended that this specific component or its equivalent should be employed. The resistors in all cases need only be of 10% tolerance, and they should be rated at $\frac{1}{2}$W, with the exception of the cathode resistor, R21 (150Ω), which should have a rating of at least 3W.

The output transformer should be capable of handling 5W. If a 3Ω impedance speaker is to be used with the unit, then a transformer ratio of 40:1 will be required; for a 15Ω speaker, the ratio is 15·5:1.

The coupling capacitors (C7 and C11) should be of

COMPONENTS LIST

Resistors:

R1 47kΩ	R7 47kΩ	R12 100kΩ	R17 100kΩ
R2 330kΩ	R8 1·5MΩ	R13 1MΩ	R18 470kΩ
R3 5·6kΩ	R9 220kΩ	R14 330Ω	R19 4·7kΩ
R4 470kΩ	R10 2·2kΩ	R15 2·2kΩ	R20 470kΩ
R5 100Ω	R11 2·2MΩ	R16 47kΩ	R21 150Ω 3W
R6 100kΩ	All ±10%, ½W carbon unless otherwise stated		

VR1 500kΩ log. potentiometer with d.p. switch (S1)
VR2 500kΩ log. potentiometer   VR3 50kΩ 1in. potentiometer

Capacitors:

C1 32μF electrolytic 350V	C9 0·05μF paper
C2 50μF electrolytic 25V	C10 16μF electrolytic 350V
C3 0·1μF paper	C11 0·02μF paper 1,000V
C4 0·05μF paper	C12 25μF electrolytic 25V
C5 16μF electrolytic 350V	C13 1μF paper
C6 25μF electrolytic 25V	C14 0·05μF paper
C7 0·02μF paper 1,000V	C15a, b 32+32μF elec. 450V
C8 25μF electrolytic 25V	C16 3000pF mica

Inductor:
L1 Smoothing choke, 10H 100mA

Transformers:
T1 Pentode output transformer. 40:1 for 3Ω speaker; 15·5:1 for 15Ω speaker
T2 Mains transformer. Secondaries 250-0-250V 70mA; 6·3V 1·5A

Valves:
V1 EF86   V2 6BR7   V3 EL84   V4 EZ80

Transistor:
TR1 XB112 or ACY20 (Mullard)

Miscellaneous:
J1 Jack socket (Bulgin type J11)
J2 Jack socket (Bulgin type J12)

Chassis, connecting wire, B9A valveholders, output socket, tagboards, etc.

1,000V d.c., and beyond suspicion as regards leakage. The valveholders, all B9A types, should, if possible, be of moulded or nylon loaded construction, to minimise the possibilities of breakdown or tracking. C5 and C10 can be the two sections of a twin-electrolytic.

It will be noticed from the circuit diagram (Fig. 1) that connections are made to the amplifier using jack plugs and sockets. Two jack sockets will be required for the inputs. One of these (J2) has integral switching contacts, which open the circuit when a plug is inserted. Thus, the effect of these contacts, when connected as shown, is to switch out the transistorised stage of the amplifier when a high impedance input is connected.

Fig. 2. An above-chassis view of the amplifier, showing the layout of the major components.
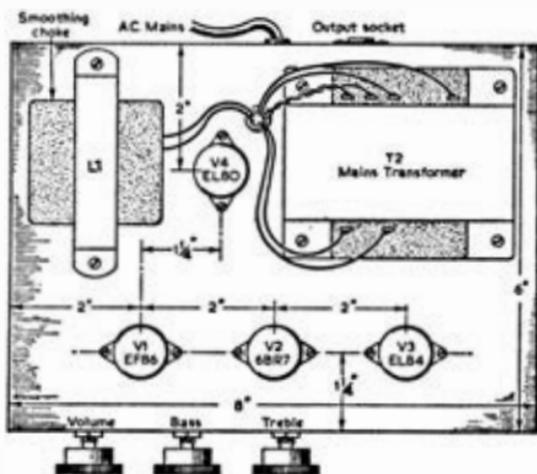
Construction

The prototype amplifier was built on an aluminium chassis, size 8 in. × 6 in. (Fig. 2). It would be impracticable to use a chassis of smaller dimensions than these. On the other hand, the use of a very much larger chassis is not to be recommended, since all connections should be kept as short as possible to reduce the effects of induction and feedback which lead to noise and instability.

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With a view to keeping down the hum level of the amplifier, all earth returns are made to an earth bus-bar, which is itself connected to the chassis at only one point (J2). This bus-bar is made from a length of 16 or 18s.w.g. tinned copper wire, which is mounted on the chassis in the manner shown in Fig. 3.

The chassis itself is earthed. If it were left "floating", i.e. not connected to earth, then trouble might be experienced due to hum.

Mounting Components

It will be seen from the diagram (Fig. 3) that the output transformer is mounted below the chassis. This transformer is mounted first, together with the mains transformer, valveholders, smoothing choke, the controls and the jack

Fig. 3. The underchassis wiring diagram.

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sockets. A point to note is that the low impedance input jack socket must be insulated from the chassis, since neither side of the low impedance input is connected to earth. In the prototype, this insulation was effected using a rubber grommet (see Fig. 4). An alternative method would be to mount the socket on a piece of paxolin or other insulating
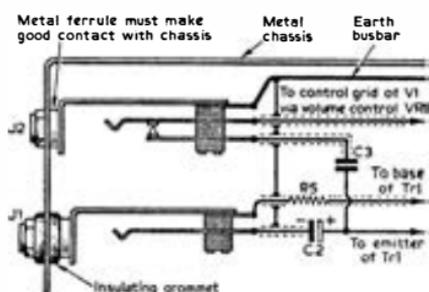

Fig. 4. Mounting details of the two jack sockets and their associated wiring.

material, and to bolt this to the chassis, taking care that no part of the jack socket touches the metal of the chassis.

Having mounted the above-mentioned components, the heater wiring should be completed. This takes the form of a pair of wires, twisted lightly together, and pressed hard against the chassis. One side of the heater wiring should be earthed. The earth bus-bar is then fixed in position.

No special precautions need be taken when wiring the amplifier, except that all connections should be kept short, and that any earth connection which may be required should be made to the earth bus-bar, and not directly to the chassis. Screened leads should be used where they are indicated in the circuit diagram (Fig. 1).

Output Sockets

In the prototype amplifier, the output was taken via a coaxial plug and socket, and coaxial cable. There is no reason, however, why any other form of socket may not be used. Since one side of the output is earthed, then a normal jack socket bolted directly to the chassis may also be used.

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Two-valve Pre-amp and Equaliser

THIS hi-fi pre-amplifier unit is suitable for use with any amplifier.

Equaliser

The circuit in Fig. 5 is focused around V1 which can be either one section of the double triode ECC83 or any high gain, double-diode-triode, such as the EBC41 or 6AT6.
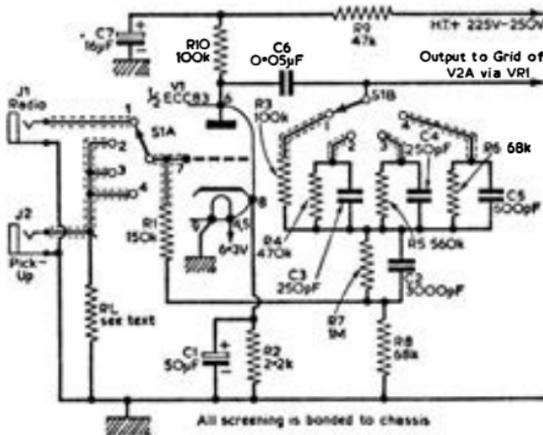

Fig. 5. The circuit of the equaliser.

The circuit of V1 provides equalisation for radio and records, as well as additional amplification for an amplifier with an input impedance of 1MΩ requiring 500mV input. Equalisation for radio is on position 1 of S1, RIAA position 2, U.S. Col. LP's position 3, and 78's position 4.

A negative feedback resistance/capacity network in the anode/grid circuit of V1 provides the necessary frequency correction. RL is not the load resistor of the pick-up, as it is in parallel with R1 + R8 and its value should, therefore, be calculated bearing this fact in mind. Adequate decoupling by C7 serves to keep hum and distortion at low level.

Fig. 6. The tone and volume control pre-amp circuit.

Tone Control System

In Fig. 6, the second stage incorporates a well-known tone control circuit with independent cut and boost in both treble and bass frequencies, as referred to a frequency of 1,000c/s.

An interesting change was the omission of C15, as a result of which a degree of current feedback was applied to V2, keeping hum and distortion at a minimum. C15 could, however, be inserted if extended high frequency response is required and where the extra gain is useful. The potentio-

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meter VR2 is centre-tapped, giving treble boost and cut at the extreme ends of its travel; VR3 likewise provides bass boost and cut.

Construction

As many constructors design their own layout, a few points only must be stressed: all control-grid wires must be screened; the tone control casings should be earthed; owing to the high gain, screening might be required, though it was not used in the prototype; all heater wires should be twisted tightly and positioned close to the chassis, as far away from the grid leads as possible.

COMPONENTS LIST

Resistors:

R1	150kΩ	R6	68kΩ 5%	R11 3·3kΩ	R16	3·3kΩ
R2	2·2kΩ	R7	1MΩ	R12 150kΩ	R17	2·2kΩ
R3	100kΩ	R8	68kΩ	R13 100kΩ	R18	100kΩ
R4	470kΩ 5%	R9	47kΩ	R14 100kΩ	R19	47kΩ
R5	560kΩ 5%	R10	100kΩ	R15 470kΩ		

All $\pm$ 10%, $\frac{1}{2}$W carbon unless otherwise stated
VR1 1MΩ carbon   VR3 500kΩ carbon
VR2 1MΩ carbon, centre tapped

Capacitors:

C1 50μF electrolytic 50V	C9 5,000pF mica or ceramic
C2 3,000pF mica or ceramic	C10 5,000pF mica or ceramic
C3 250pF silver mica 5%	C11 140pF silver mica 5%
C4 250pF silver mica 5%	C12 0·1μF paper
C5 600pF silver mica 5%	C13 32μF electrolytic 450V
C6 0·05μF paper	C14 25μF electrolytic 450V
C7 16μF electrolytic 450V	C15 50μF electrolytic 50V
C8 0·1μF paper	C16 0·05μF paper

Miscellaneous:

V1 ECC83 or 12AX7	V2 ECC83 or 12AX7
J1 Jack socket	J2 Jack socket
S1 2-pole, 4-way rotary switch	

Two B9A valveholders. Piece of aluminium 10 in. $\times$ 9 in. for the chassis. Connecting wire, screened wire, etc.
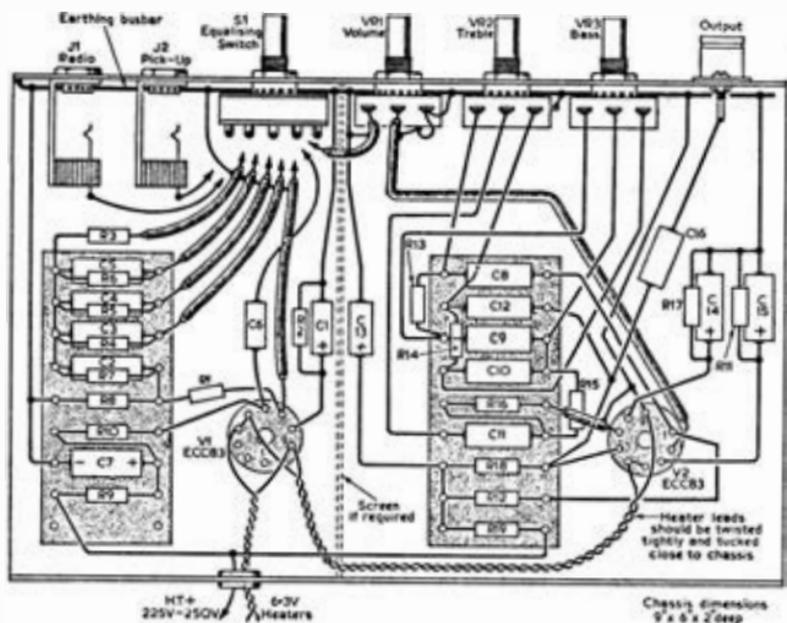

Fig. 7. The underchassis wiring diagram of the combined unit.

Construction on group boards will give a neat appearance and facilitate servicing. It should be remembered that the layout of components can make or mar the capabilities of the pre-amp.

The lead going to R11 should be kept close to chassis since the absence of C15 (if, in fact, it is omitted) may cause induced hum.

All earthing must be made to a bus-bar, earthed as close to the input sockets as possible, and there only.

Best results would be obtained if C3, C4, C5, C11, R4, R5, R6 were of close tolerance, i.e. 5%.

Component layout

For those who wish to follow the prototype, the layout of components is given in Fig. 7.

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Mains Gramophone Amplifier

THIS small amplifier costs very little to build—in fact, many constructors will probably be able to find most of the parts in the spares box. Nevertheless, it has quite a rewarding performance and a gain which is adequate for most types of pick-up, radio feeder unit, etc.

Circuit

The very simple circuit is shown in Fig. 8. The input signal is applied to a volume control, VR1 and thence via the tone control network VR2/C1, to the grid of a resistance-coupled pentode voltage amplifier, 6J7. The resistor R3 is the anode load and the screen is fed from a potential divider across the h.t. supply. Normally, the screen of a resistance-coupled pentode must be decoupled by a fairly large capacitance, but if the impedance from screen to cathode can be made reasonably low, as it is in this case, little is lost by omitting the decoupling.

The cathode of the valve is tapped on to the potential divider at a lower point which is selected so that the current in the potential divider together with the cathode current of

the valve, produce the correct bias across the parallel-connected resistors R4 and R5. These in turn are selected so that their relative values in series cause the desired amount of negative voltage feedback from the output transformer secondary to be injected at the cathode of V1. It will be noted that V1 operates with both current and voltage feedback so that its input impedance is high and only a small capacitance is needed in the tone control network.

Output stage

The signal from V1 is transferred via C2 to the grid of V2, which is an output tetrode, 6V6. R6 is the grid resistor and R7 provides cathode bias. It will be seen that R7 is not bypassed so that V2 also operates with negative current feedback. This is undesirable as a rule, because it has the effect of increasing the apparent anode resistance of the valve so that its less desirable qualities are accentuated. In this case, however, this effect is counteracted by the negative voltage feedback and the net result is a reduction in the
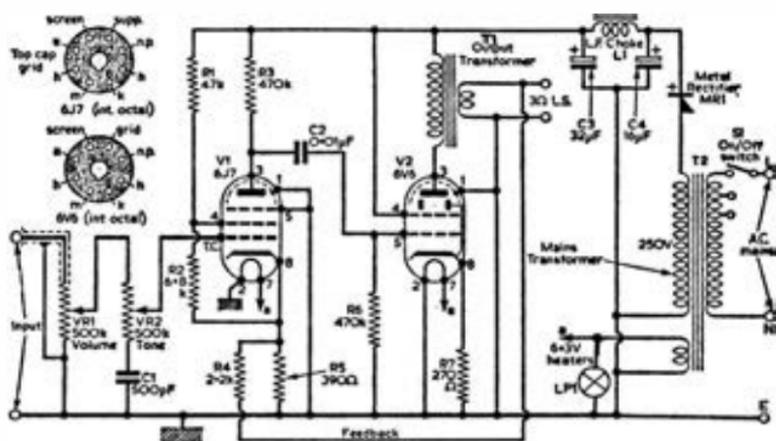
Fig. 8. The circuit. The inset diagram gives details of the valve bases.

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apparent anode resistance and an improvement in the linearity of the amplifier.

The ouput transformer in the anode circuit should have a ratio of about 40:1 to match a 3Ω speaker to the optimum load for the valve which is 5,000Ω. Only an inexpensive transformer is required as its deficiencies will be largely made good by negative feedback.

Power Supply

A double-wound mains transformer is used, thus providing complete isolation of the amplifier from the mains. It should have a 250V half-wave secondary capable of supplying 50mA and a 6·3V 1A winding for the valve heaters and indicator lamp; the latter should not consume more than 0·25A if the heater current is to be kept down to 1A. Any type of half-wave metal rectifier can be used so long as it can pass 50mA at 250V. Smoothing is provided by the electrolytic capacitors C3 and C4 in conjunction with the choke L1 and, as the hum is very well taken care of by negative feedback, it is not necessary that the choke should have a large inductance.

Construction

There is nothing critical about the layout and the amplifier can be constructed in any form suitable for the purpose to which it is to be put. The prototype was built to fit into a commercial record player cabinet of conventional shape, having a vertical section to the left of the turntable measuring about 12 in. × 9 in. × 3 in. and this form will be described.

The speaker is mounted on a piece of plywood or hardboard 11 in. × 6¾ in. and the amplifier chassis which is 2½ in. wide, is secured to the hardboard by 4B.A. bolts, two at each side and two at the bottom.
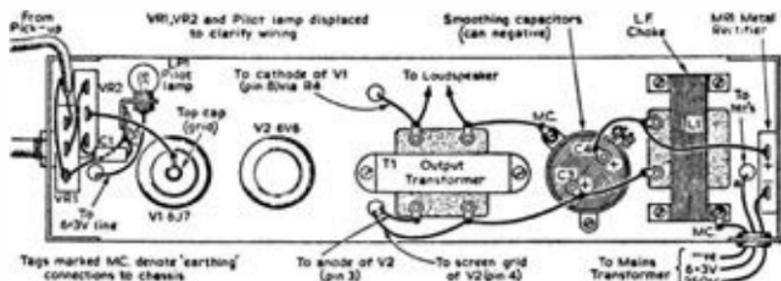
The chassis is of 18-gauge aluminium sheet to the

measurements given in Fig. 9.

After drilling, the two end sections of the chassis are bent up perpendicular to the centre section or base of the chassis, forming a "U", 11 in. × 6 in. × 2½ in. wide (Fig. 9). The ¾ in. flanges on the side sections are bent inward and those on the centre section bent down forming the plinth of the amplifier.

The potentiometers VR1 and VR2 and the pilot lamp are mounted on one of the side sections and the metal rectifier MR1 is fitted on to the other. All the other components (except for S1 and the mains transformer) are mounted above or below the base section (Fig. 10).

The mains transformer is mounted separately in the player cabinet underneath the motor board, remote from the pick-up, and its output is taken to the chassis at a point adjacent to MR1 by a 3-core cable—h.t., l.t. and common negative. The transformer can thus be of any reasonable shape and

Fig. 9. Dimensions of the chassis before it is bent as indicated on the diagram, to form a "U" shape.

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Fig. 10. The chassis layout and wiring diagram.

size and there will be no interaction between it and other components. The switch S1 is fitted to the cabinet adjacent to the point of entry of the mains lead.
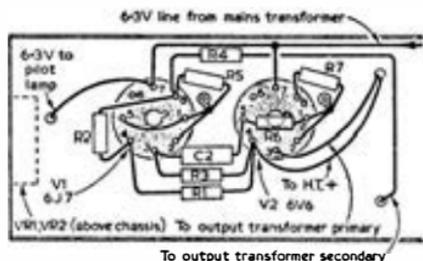
Fig. 11. Wiring under the chassis which is wholly associated with the two valve-holders.

Components and Wiring

All the components are standard and there is plenty of room for them in the chassis—in fact, there is considerable scope for miniaturisation to suit smaller cabinets. Of the resistors, R1 and R7 should be 1W and the remainder $\frac{1}{2}$W or $\frac{1}{4}$W. The capacitors should be 350V. If valves other than the specified types are used, they should have the same general characteristics or some alteration may be required in the feedback loop. Wiring diagrams are given in Figs. 10 and 11. Tinned copper wire of 22 s.w.g. is suitable for all the wiring and lengths of more than an inch or so should be covered with sleeving.

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COMPONENTS LIST

Resistors:
R1 47kΩ 1W   R3 470kΩ   R5 390Ω   R7 270Ω 1W
R2 6·8kΩ   R4 2·2kΩ   R6 470kΩ
All ± 10%, ½W carbon unless otherwise stated

VR1 500kΩ carbon track   VR2 500kΩ carbon track

Capacitors:
C1 500pF ceramic   C3 32μF electrolytic 350V
C2 0·01 μF paper   C4 16μF electrolytic 350V

Miscellaneous:
L1 Smoothing choke 10H 50mA
LP1 Lamp, 6·3V 0·15A
MR1 Metal rectifier, contact cooled, 350V, 50mA
T1 Pentode output transformer, about 40:1 for 3Ω loud-
   speaker
T2 Mains transformer. Secondaries: 0-250V 50mA;
   6·3V 1A
S1 S.P.S.T. toggle switch 250V 1A
V1 6J7, KTZ63 or Z63   V2 6V6, KT61 or EL33

Operation

When construction is complete, a check should be made
with a meter between C4 and chassis to see that there are no
shorts in the h.t. wiring. The power can then be switched
on and the h.t. voltage checked. If there is instability it will
be due to incorrect connection of the voltage feedback
circuit and will disappear if the connections to the output
transformer primary are reversed.

The input arrangements to V1 may, of course, be modi-
fied to suit the signal source as desired; the performance
will not be affected except insofar as the response may be
modified by alterations to the tone control network.

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Some Common Causes
of Distorted Output

RADIOGRAMS, commercial or home-made, can suffer
from a number of troubles in addition to the usual
radio receiver faults with symptoms peculiar to
themselves and it is as well for all audio enthusiasts to be
acquainted with the most common ones, especially when
designing or building their own equipment. These defects
are generally either pick-up faults or turntable faults which
can also apply to simple record players.

Pick-up Faults—Hum

When present, hum is more pronounced on radiograms,
because of the larger speakers fitted and the extra attention
given to the audio circuit by the designer to increase bass
response. The larger baffle area for the speaker also accentu-
ates this. Never attempt to clear hum troubles on a gram
chassis without the accompanying speaker connected and
fitted in its cabinet. If this is not possible, then make sure
that the bench speaker used is well baffled, because a low-

level hum which is tolerable on a small speaker turns out to be overpowering when the chassis is fitted back in its cabinet.

Hum that occurs when the gram is switched to records, and is unaffected by the volume control, is residual and is introduced in the circuit after the volume control. Deteriorating smoothing capacitors and partial heater-to-cathode leaks on following audio valves are the most likely reasons.

Hum which reduces with the volume control setting on "gram" only, nearly always originates from the pick-up.
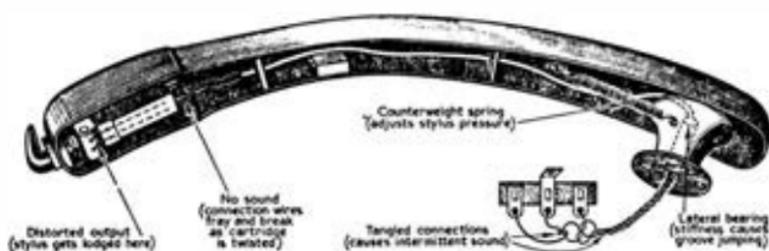
Fig. 12. Trouble spots on a typical pick-up arm.

This can be checked by unplugging the pick-up leads from the chassis when the hum drops to a tolerable level. Check that the pick-up leads are not reversed—the screen should be earthed or connected to the chassis—this is a common trouble as many commercial grams are not provided with polarised sockets and the plugs can be fitted the wrong way round by a novice, without affecting the sound but causing hum.

Variation in hum level when the pick-up or motor board are handled, denotes incorrect earthing. Check the earth connections under the board but *do not add extra earths to chassis in the case of a.c./d.c. grams* or the motor board might become live to mains. Some a.c./d.c. grams give less hum with the mains plug a certain way round in the supply socket (taking the chassis to neutral).

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Other Pick-up Faults

Distortion on gram only can be caused by the stylus or by the pick-up crystal itself (see Fig. 12). The stylus can be checked on rotating cartridges by reversing to the other stylus; if this gives undistorted results (on the correct speed record), then the other stylus is proved defective. In any case, it pays to replace worn styli before evidence of wear is audible. See that the stylus is properly located on the crystal transmission pad. Too often, through mishandling and heavy set-down on auto-changers, the stylus becomes dislodged and is found jammed between the cartridge side and pad (Fig. 12). Debris picked up as dust should be cleaned periodically from the stylus.

Cracked Crystals

Good reproduction on radio but weak distorted sound from records is often due to a cracked crystal. This can be checked by unplugging the pick-up plugs from the chassis and fitting a crystal microphone or a substitute crystal pick-up from another record player. Good results from any of these shows the original crystal to be at fault. Crystal cartridges are fragile and can be fractured by careless handling—it is not generally known however that they are prone to failure with high temperature, and portable mains radiograms with upper vents discharging heat into the space under the lid, "cook" pick-ups if the gram is operated for an hour or two with the lid down.

One of the commonest troubles with turnover pick-up cartridges is the breaking off of the fine wire leads at the terminals on the cartridge due to the constant flexing with rotation. *Do not resolder direct to the terminals* but remove the clips first and solder to them, otherwise the heat may damage the crystal. This trouble is sometimes intermittent if the broken lead makes and breaks with vibration.

Insufficient pick-up weight can also cause cutting out of the sound owing to the stylus tip being held clear of the bottom of the groove. An adjustable lowering stop at the

Fig. 13. Wear on a stylus which can be caused if the balance of the arm is incorrectly adjusted.

base of the pick-up can allow the stylus to drop lower, or if the arm is almost weightless, the counter balance spring also situated under the arm at the base, can be slackened off one hole. Too much weight however causes excessive wear of records and styli (see Fig. 13).

Fig. 14. Mechanical vibrations in a radiogram cabinet can set up an acoustic feedback loop, resulting in "howl".
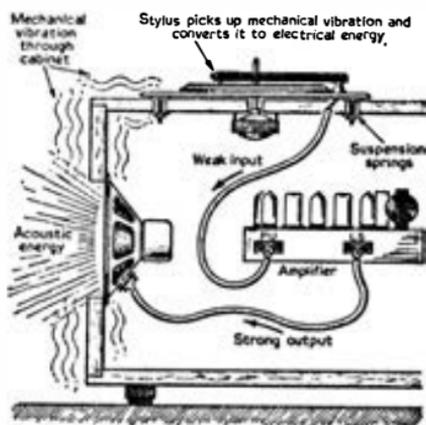
Microphony

Howling which commences when the volume is increased above a certain minimum (similar to amplifier acoustic feedback in a hall) is due to the cabinet conducting vibrations from the loudspeaker to the turntable and record where they are picked up by the stylus, amplified, and sent round the feedback loop again (Fig. 14). This is usually a fault with home constructed equipment where the motor board is

unsprung or badly fitted. A rigid cabinet with baffle bracing reduces mechanical vibration and raises volume at which the threshold of howl is reached. On manufactured equipment this trouble is more often due to tight, unslackened, turntable transit screws.

Turntable Speed

The usual fault causing wow, slurring, and slowing of the record is not the turntable at all but the record slipping and skidding on its label with the one in contact with it (Fig. 15). This loss of friction can cause a warbling noticed on sustained high notes or severe slowing and at times stopping
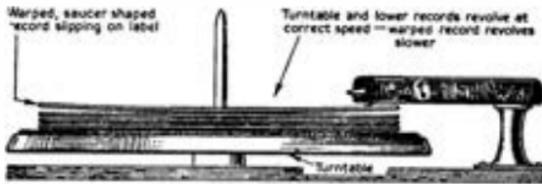
Fig. 15. Slurring and wow can sometimes be caused by nothing more than a warped record

of the top record while the underside pile continue to rotate normally. The upper record when examined will be found to be saucer-shaped or to possess a small eruption on the label lifting the upper record clear of the main friction plane. A greasy or polished rim on the jockey wheel can cause the same effects but this time the turntable will be found to be turning slowly (Fig. 16). Check the pressure

Fig. 16. Some likely causes of permanent slow speed.
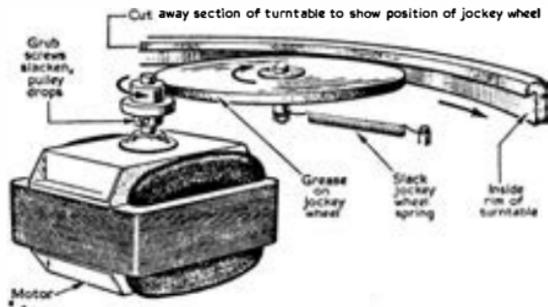
spring pulling the jockey wheel on to the pulley for reduced tension.

Permanent, steady slow speed is most usually caused by engagement of the wrong pulley. The pulley bush can slip down the motor spindle if the grub screws loosen accidentally and with little flywheel momentum to iron it out, wow or flutter may be present.

Generally, quiet passages or sustained notes show up this effect which is heard as a flutter or gnawing sound. It may be due to a fault in the rotor causing uneven pull, or to a flat on the drive pulley.

Groove Jumping

Microgroove records in good condition that exhibit this trouble at any random place across the record, point to excessive lateral stiffness of the pick-up arm. See that the pick-up screened wire is flexing freely without resistance and that the central bearing holding the arm is lightly lubricated.

Groove jumping that takes place always near the end of the record is often caused by the trip mechanism becoming stiff. When examining this small, lightly riveted pawl, under the turntable, *do not oil it*. Clean it and exercise it with a few twists with the fingers to clear any chemical corrosion that can increase friction, but to oil it will only attract the dirt and dust and cause the same trouble later.

Output Transformers and Loudspeakers

L OUDSPEAKER faults can be divided into two primary classes, mechanical and electrical. An electrical fault usually causes either complete failure or intermittent operation, while a mechanical fault can produce a diversity of symptoms. Some of these faults may not immediately be associated with the loudspeaker, but may be put down to distortion in the audio-amplifier.

Electrical Faults

Electrical faults are few. In essentials, the modern loudspeaker simply comprises a coil of wire, called the speech coil, which is arranged to move in a magnetic field. The output transformer is sometimes included on the loudspeaker chassis, and for this reason the transformer will be considered as part of the loudspeaker in this chapter. It should be noted, however, that the output transformer may not, in fact, be on the loudspeaker itself, but on the chassis of the amplifier.

The transformer serves to match the impedance of the

speech coil to the output valve, and thus has two windings; a low resistance winding which is connected across the loudspeaker's speech coil, and a high resistance winding which is connected to the anode circuit of the output valve. The transformer has a step-down ratio from the output valve to the speech coil, and this ratio is computed in relation to the impedances of the anode circuit and the speech coil.

Fig. 17 gives the basic loudspeaker circuit, and it is fairly obvious that there are four main possibilities of failure. These are a break in the speech coil winding, a break in one of the windings of the transformer, a disconnected or broken wire in the loudspeaker circuit and, possibly less obvious, shorting turns in the output transformer.

Fig. 17. Typical receiver output stage.
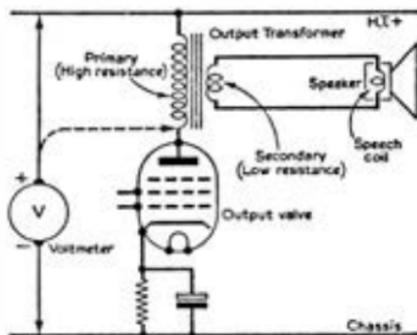
Complete Failure

The first hint that the loudspeaker may be responsible for the failure of an amplifier is complete absence of the residual mains hum which can normally be detected by holding an ear close to the loudspeaker. In this event, the speaker should be unplugged from the transformer and the speech coil connected across a 1·5V battery in series with a resistor of about 100Ω. The resulting 15mA of current in the speech coil should produce quite loud clicks and crackles as the battery connection is scraped (see Fig. 18). If this happens, then the speaker can safely be said to be electrically

sound. The resistor is included to limit the current, for without it, a relatively high current can flow in low resistance speech coils and damage the winding on low-power units.

If the speaker remains dead, then there is a break somewhere in the speech coil circuit. Very flexible wire is used to connect the speaker terminals to the speech coil and a break sometimes occurs at one of the terminations, which can usually be mended by skilful soldering. If these wires are in order, a check should be made of the soldered connections or small wire tags in the centre of the cone which anchor the ends of the speech coil. Although these wires are held by cement, they can be traced to the soldered blobs with a pin or pointed tweezers. If the break is located, the wire should be freed from the cement, cleaned of enamel, resoldered and finally recemented in position to avoid vibration.

If there is no break at these points, the trouble lies in the speech coil winding and, apart from replacing the cone and speech coil assembly complete, there is little that can usually be done.

Fig. 18 (left). A method of checking loudspeakers.
Fig. 19 (right). An output stage incorporating an internal speaker switch; operated when an extension speaker is used.
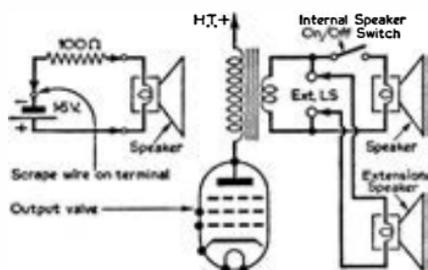
If the test given in Fig. 18 produced crackles, the wires connecting the loudspeaker unit to the secondary of the output transformer should be checked for continuity. Some amplifiers incorporate a switch in the speech coil circuit to switch on and off the internal speaker if an extension speaker

is connected, as shown in Fig. 19. The switch may simply be a plug and socket arrangement or something more elaborate, but in either case it should be checked for continuity.

Output Transformer Test

There are two methods of checking the output transformer complete with speaker. One is to connect a voltmeter between the anode of the output valve and the chassis of the amplifier, after first checking that h.t. voltage is in fact present on the h.t. line, as shown in Fig. 17. If the primary of the transformer is in order, a voltage slightly lower than the h.t. line voltage will be registered at the anode of the valve (assuming that the anode circuit is taking current). If the speaker circuit is normal, slight crackles should be heard from the loudspeaker as the voltmeter test probe is scraped on the anode tag. An alternative method is to disconnect the primary of the transformer from the valve anode and h.t. line and connect the winding across a 1·5V battery. If all is in order, crackles should again be heard from the loudspeaker as the battery connection is scraped.

Transformer Turns Ratio

The turns ratio of the output transformer is governed by the "optimum load" of the output valve and the impedance of the speech coil, both in ohms. Optimum load values are given in valve data booklets and the speech coil impedance is sometimes marked on the loudspeaker chassis. It should be noted that the impedance is generally taken at 1,000c/s, and is, therefore, slightly higher numerically than the d.c. resistance of the speech coil. As a rough guide, the imped-ance is a little over twice the d.c. resistance value.

The turns ratio is equal to the square root of the optimum load divided by the speech coil impedance which, put in

mathematical terms is:

$$Tr = \sqrt{(Lo/Zs)}$$

where Tr is the turns ratio, Lo the optimum load and Zs the speech coil impedance. The expression may be altered to find Lo or Zs.

For example,

$$Lo = Tr^2 \times Zs,$$
$$\text{and } Zs = Lo/Tr^2.$$

Thus, given any two factors, the third or unknown can always be found.

If, for instance, the optimum load were 2,000Ω and the speech coil impedance 5Ω, then 5 divided into 2,000 is 400, and the square-root of 400 is 20. The required transformer in that case would have a turns ratio of 20:1, which is a fairly common value.

The same reasoning applies to any loudspeaker matching problem. Some transformers have tapped primaries or secondaries or both to enable the best match to be obtained from any optimum load to any speech coil impedance. Tapped transformers of a similar nature are also available to match a low impedance loudspeaker into a low impedance source. For example, a matching transformer would be required to match, say, a 15Ω loudspeaker to a 3Ω extension loudspeaker socket on a radio or radiogram, or vice versa.

Checking the Turns Ratio

An approximate idea of the turns ratio of an output transformer can be obtained by applying low voltage a.c. across the secondary winding (low resistance winding) and measuring the voltage appearing across the primary winding, as shown in Fig. 20. About 2V a.c. should be used, and this can be obtained from a step-down mains trans-

former. It should be measured as accurately as possible on an a.c. voltmeter, as also should the voltage across the primary winding. The ratio of the two voltages is approximately equal to the turns ratio of the transformer. For example, if 2V were applied and 60V appeared across the primary winding, then the turns ratio would be 60:2, or 30:1.
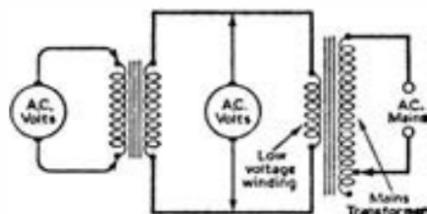

Fig. 20. Determining the turns ratio of an output transformer.

Shorting Turns

An output transformer with shorting turns may not cause total failure of the amplifier, but will most certainly cause low volume and distortion, with loss of bass frequencies. By using the set-up shown in Fig. 20, shorting turns will give an obviously low output voltage and the transformer will quickly heat up.

Mechanical Faults

Mechanical faults rarely cause total failure of the set, but they produce noises, such as buzzes, rattles and distorted reproduction. A distinct buzz may be caused by the speech coil being out of centre and touching the magnet pole pieces or by a damaged or broken centring device, a split or tear in the cone, loose turns of wire on the speech coil, loose mounting bolts holding the loudspeaker in the cabinet, vibrating connecting leads, etc.

Such faults can usually be located by operating the set with the loudspeaker in its normal position and touching the various parts of the loudspeaker in an endeavour to pin-point the source of the noise. If the speech coil is out

of centre, lightly pressing a finger at a certain point on the cone will often either clear the effect or, at least modify it.

Eventually, it will be necessary to remove the loudspeaker from the cabinet in order to effect a permanent repair. If the fault is due to the speech coil being out of centre and the centring device has not shifted, the centring "spider" may have come loose from the cone or the speech coil former may have warped. A centring device which has cracked or parted from the cone can usually be repaired without difficulty, and repairs have been carried out successfully when the speech coil has actually broken away from the centre of the cone.

The cone and speech coil assembly should be removed from the loudspeaker chassis and a cellulose cement should be applied between the coil former and the cone and allowed to set hard before reassembling. Loose turns on the speech coil can also be locked with a similar cement applied in a very thin layer.

If the speech coil former has warped, there is usually little that can be done, for even if the former is squeezed back to approximate its original shape, it very often warps again as soon as it warms up in the amplifier. (This trouble, in fact, often happens in small mains-type radio receivers, which become very hot inside and use a small loudspeaker. When the set is first switched on the reproduction may be normal, but after a warm-up period loudspeaker distortion becomes apparent, owing to warping of the speech coil). The only cure here is to replace the loudspeaker.

Speaker Centring

To re-centre a loudspeaker, the retaining screw or screws of the centring device should be slackened. Pressure should then be applied to the cone as required and the retaining screw or screws tightened. The process may have to be repeated several times in order to achieve the best results.

The speech coil must, of course, operate within the magnet gap without touching the pole pieces.

Gauges for pushing between the coil former and the magnet to hold the former in the correct position while tightening the centring screws are available, but if these are used care should be taken to avoid using too much pressure, otherwise the former may spring out of centre when the gauges are removed.

One way of checking whether the speech coil is correctly centred is to apply about 2V a.c. via a 100Ω current limiting resistor, to the speech coil terminals on the loudspeaker chassis. If all is well, the cone will vibrate freely without harshness or obvious fouling. The correct centring position can be established, if necessary, by adjusting the centring device with the loudspeaker connected to the a.c. supply, but the speech coil should not be allowed to rub on the pole pieces for any length of time, otherwise the coil winding may be damaged.

Metallic particles of a ferrous nature, or even rust, sometimes reduce the magnetic gap or stick to the pole pieces. These can be removed by pressing a piece of Plasticine or soft wax on the pole pieces and magnet assembly.

Crossover Networks for Loudspeakers

PROPRIETARY loudspeaker crossover networks are fairly expensive items, and are not always available with the required impedance and crossover frequency. This chapter gives details of how to design and make crossover networks for any impedance and frequency.

Basic Theory

Loudspeaker crossover networks are in fact perfectly straightforward prototype (or constant-K) filter whole-sections or half-sections. The mathematical theory of these filters may be found in any standard text-book and it is not proposed to deal with it here.

Two types of filter are used in these networks. One is a low pass filter, which, as the name would suggest, passes only frequencies *below* its cut-off frequency. Signals above this frequency are subjected to attenuation and the amount of attenuation increases with frequency. The circuit configuration of such a filter is shown in Fig. 21 (a) and (b), where both a whole- and a half-section are depicted. The second

type of filter is a high pass filter—Fig. 21 (c) and (d)—which passes without attenuation only those frequencies *above* its cut-off frequency, frequencies below being subjected to attenuation. In this case, the degree of attenuation increases as the signal frequency decreases.

Attenuation

If a prototype low pass filter section is designed to have a certain cut-off frequency, it does not start to attenuate

Fig. 21. (a) *Low pass filter—whole-section;* (b) *low pass filter—half-section;* (c) *high pass filter—whole-section;* (d) *high pass filter—half-section.*
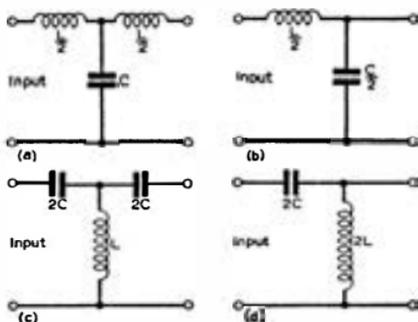
sharply at that chosen frequency. In practice, it will be about 20% above the cut-off frequency before 6dB of attenuation is reached, though at a frequency 20% below cut-off frequency, the filter will introduce an attentuation of about $1\frac{1}{4}$dB. This performance is typical of prototype filters. Filter sections can be designed to have a very rapid attenuation if so desired, but such filters lie outside the scope of this book.

If it is desired to feed a high frequency loudspeaker or "tweeter", this may be done in one of three ways (see Fig. 22); (a) it can be connected in series with a single capacitor of suitable value; (b) it can be fed via a high pass half-section filter; (c) it can be fed via a high pass whole-section filter.

Method (a) is not normally considered satisfactory since the rate of attenuation below the crossover frequency is

insufficient. Both methods (b) and (c) are acceptable, though method (b) is, in fact, the method normally used.

The rates of attenuation below cut-off frequency, per octave ($2 \cdot 1$ frequency change), are:

        (a) capacitor feed   6dB/octave
        (b) half-section filter   12dB/octave
        (c) whole-section filter 18dB/octave

Design Data

Prototype filter sections are designed from the following equations

    Low Pass   High Pass

$$fc = \frac{1}{\pi\sqrt{(LC)}} \qquad\qquad fc = \frac{1}{4\pi\sqrt{(LC)}}$$

$$Z = \sqrt{\left(\frac{L}{C}\right)} \qquad\qquad Z = \sqrt{\left(\frac{L}{C}\right)}$$

where fc is the cut-off frequency in cycles per second, Z is the impedance of the filters in ohms and L and C are the

Fig. 22. Three methods of feeding a "tweeter".
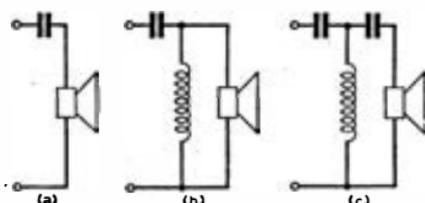
values of inductance and capacitance in henrys and farads.

To save the necessity for long calculations, these formulæ have been re-arranged and partly worked out as below:

    Low Pass   High Pass

$$L = \frac{318Z}{f}\mu H \qquad\qquad L = \frac{79 \cdot 5Z}{f}\mu H$$

$$C = \frac{318}{fZ}\mu F \qquad\qquad C = \frac{79 \cdot 5}{fZ}\mu F$$

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where f is the desired crossover frequency in kc/s and Z is the loudspeaker speech coil impedance in ohms.

Example

Designing a loudspeaker crossover network with a crossover frequency of 5kc/s with a loudspeaker speech coil impedance is 15Ω.

$$\text{Low Pass}$$
$$L = \frac{318Z}{f} = \frac{318 \times 15}{5} = 954\mu H$$
$$C = \frac{318}{fZ} = \frac{318}{5 \times 15} = 4 \cdot 24\mu F$$

$$\text{High Pass}$$
$$L = \frac{79 \cdot 5Z}{f} = \frac{79 \cdot 5 \times 15}{5} = 238 \cdot 5\mu H$$
$$C = \frac{79 \cdot 5}{fZ} = \frac{79 \cdot 5}{5 \times 15} = 1 \cdot 06\mu F$$

The actual values of the components required for the half-section filters will therefore be:
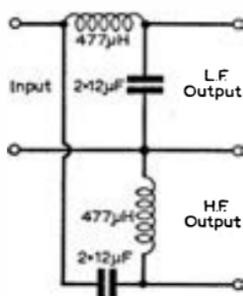
Low Pass
Inductor ($\frac{1}{2}$L) $477\mu H$
Capacitor ($\frac{1}{2}$C) $2 \cdot 12\mu F$

High Pass
Inductor (2L) $477\mu H$
Capacitor (2C) $2 \cdot 12\mu F$

Fig. 23. A 15Ω, 5kc/s crossover network.

In other words—the components needed for the high and low pass half-sections are the same. It should be noted that will not be the case if whole-section filters are used. The

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circuit for the network designed above is shown in Fig. 23. Table 1 gives component values required for certain specific frequencies for 3 and 15Ω loudspeakers.

TABLE I

Impedance	3Ω		15Ω	
Frequency	Inductor	Capacitor	Inductor	Capacitor
0·5 kc/s	954μH	106μF	4780μH	21·2μF
1·0 kc/s	477μH	53μF	2390μH	10·6μF
2·0 kc/s	238·5μH	26·5μF	1195μH	5·3μF
5·0 kc/s	95·4μH	10·5μF	478μH	2·12μF
10·0 kc/s	47·7μH	5·3μF	239μH	1·06μF

Construction

When the component values for the desired network have been arrived at, all that remains is to make the unit.

The capacitors should preferably be paper or metallised paper types. However, certain of the values listed in Table 1 preclude the use of such capacitors, and resort must therefore be made to electrolytic capacitors. These may be either of the reversible type or of the uni-directional type. If the latter are employed their working voltage should be not less than 25 times the maximum a.c. voltage to be applied to them. Thus for a 15W amplifier using 15Ω loudspeakers, the capacitors should have a working voltage of not less than $25 \times 15 = 375$, whilst for a 3W amplifier using 3Ω loudspeakers the voltage required is only $25 \times 3 = 75$.

Performance

A 5kc/s crossover network was made up and the perform-

ance measured both with half-section filters and with whole-section filters. The results of this test are shown in Fig. 24. The greater attenuation of the whole-section filters can be seen clearly.
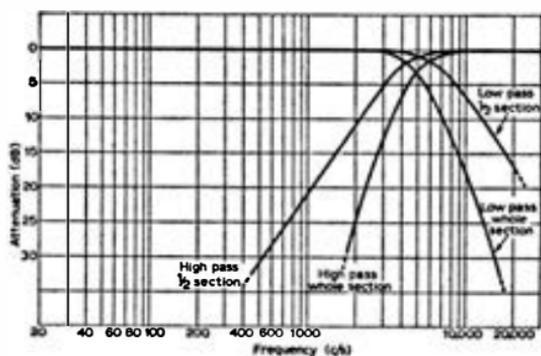
Tolerances

It is felt that it should be pointed out here that if strictly accurate results are to be obtained, the actual capacity of the capacitors employed in crossover networks, whether paper or electrolytic, should be close to their calculated value. Since paper capacitors have typical tolerances of $\pm 20\%$, and electrolytics tolerances of $-20 + 100\%$, this is not easy to ensure. Many capacitors when measured, do, in fact, have values of capacitance which are very much closer to their nominal value than their tolerances allow.

To ensure complete success, the exact value of the capacitors should be measured on a capacity bridge. However, if a bridge is not available *paper* capacitors (not electrolytics) can be measured with sufficient accuracy by the following means. Connect the unknown capacitor in series with a variable resistor and connect the series combination to a 50c/s a.c. supply, the voltage of which does not exceed $\frac{2}{3}$ of the d.c. working voltage of the capacitor. Adjust the series resistor until the voltage measured across

the capacitor is the same as that measured across the resistor. The capacity of the capacitor is given by $3180/R\,\mu F$, where R is the value of the resistor in ohms. The voltmeter used for this test should be of not less than $500\Omega/V$ on a.c.

The inductors, fortunately, present little difficulty. For ease of construction the coils are wound on a 3 in. diameter former using 22 s.w.g. enamelled wire. The coils are "random wound" over a length of about $\frac{1}{2}$ in., and when the required number of turns has been wound, the coil is slipped off the coil former and bound together with insulation or other suitable tape. For coils made as described above, an inductance of $340\,\mu H$ will be obtained with a coil of 50 turns. The number of turns required for a given inductance, $X\,\mu H$, may be calculated from the expression $N = 50\sqrt{\dfrac{X}{340}}$ turns. The quantity of wire required may be calculated from the fact that a coil of 50 turns weighs $1\frac{7}{8}$ oz.

These coils, being air-cored have an inductance which is independent of current, and cannot, unlike iron-cored coils, introduce distortion.

Coil spacing

When the capacitors and inductors have been obtained, it only remains to make a suitable assembly and wire it together. The inductors should be spaced at least 3 in. from one another and should be kept 2-3 in. away from any large metal object. If the coils are at right angles to one another, then there is no limitation on their spacing.

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Seven-watt Quality Amplifier

QUALITY reproduction is commonly associated with large power valves, expensive transformers, etc., but if fundamental principles are observed, excellent results can be obtained with simple circuits and inexpensive components. The amplifier described here was designed for use with a v.h.f. tuner; it has a frequency range which is more than adequate for the purpose and a silent background which does justice to the quality of the transmissions.

Circuit

As will be seen from Fig. 25, the output stage employs two pentodes, EL42s which in class AB1 push-pull operation will produce 7W. This is more than enough for the average listener and the full output is unlikely ever to be required. The valves were selected because, although they are slightly more expensive than some alternatives, they have a comparatively small current requirement which enables the power supply to be reduced in bulk, weight and

cost. They are operated with 250V on anodes and screens and share a common bias resistor. This does not have to be bypassed, since the signal currents cancel out at the cathodes if the valves are reasonably well matched. A word of warning here—the amplifier must not be switched on unless *both* output valves are in place.

Output Stage

The output transformer should have a reasonably high primary inductance and, to match the anode-to-anode load, which is 15kΩ, should provide a ratio of about 70 to 1 for a 3Ω speaker or 35 to 1 for a 15Ω type. The primary inductance of the transformer used in the prototype is 45H
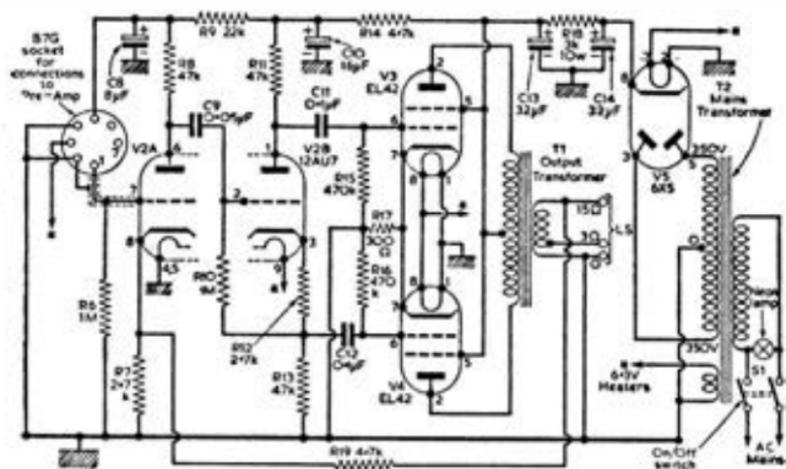
Fig. 25. The main amplifier circuit.

with no d.c. An expensive component is not necessary; indeed the extended frequency response obtained from one may introduce phase shift leading to instability when negative feedback is applied. The circuit relies on negative feedback to make good the shortcomings of the cheaper transformer.

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Inverter

The output stage requires a signal of about 35V peak grid-to-grid for full power, and this is provided by one half of the double triode valve, 12AU7, connected as a phase inverter. The circuit shown is simple but linear in the extreme, owing to heavy negative current feedback in the cathode load resistor. This feedback raises the input impedance of the stage to about ten times the value of the grid leak, i.e. 10MΩ. Quite a small coupling capacitor from the previous stage would therefore be adequate for low frequency response, but it is not advisable to reduce it below $0.05\mu F$ or the high impedance from the grid of V2B to earth will make the grid unduly susceptible to electrostatic hum pick-up. For the same reason, the anode load in the preceding stage should not be too high. The cathode bias resistor, R12 is small in relation to R13, so that there is no point in bypassing it.

A.F. Stage

The phase inverter is fed from the other half of the 12AU7, arranged as a conventional resistance coupled amplifier with an anode load of 47kΩ. Decoupling and additional smoothing are provided by C8 and R9 and the linearity of the stage is improved by omitting the usual bypass capacitor from the cathode circuit. This omission also provides a suitable point for the injection of negative voltage feedback from the output transformer and of course, saves the cost of the capacitor.

The resistors R7 and R19 are selected for two purposes; their value in parallel provides the correct bias for V2A and their relative values determine the proportion of the output voltage to be fed back to the valve cathode. The valve, of course, operates with both current and voltage feedback.

Tone Controls

Tone controls are not strictly necessary for reproduction of v.h.f. programmes, but comprehensive controls have been provided so that the response can be adjusted to individual taste. The gain from the grid of V2A onwards is quite low and the controls are inserted at this point (Fig. 26) so that they shall not be sensitive to hum.

VR2 is the treble control and VR3 the bass control.

The values of the associated capacitors C4, C5, C6 and C7 can be varied to taste; those shown gave a wide range of control in the prototype and should be generally satisfactory. No special screening is necessary.

Pre-amplifier

The pre-amplifier stage VI employs a resistance-coupled pentode, 6AM6 (Fig. 26). This gives the amplifier more overall gain than is necessary for radio reproduction, but makes it suitable for use with other lower level inputs. The screen of this valve requires to be decoupled by not less than $0.5\mu F$, and as a paper capacitor of this value is rather bulky, a $2\mu F$ electrolytic is used. The cathode

Fig. 26. Circuit of the pre-amplifier section.

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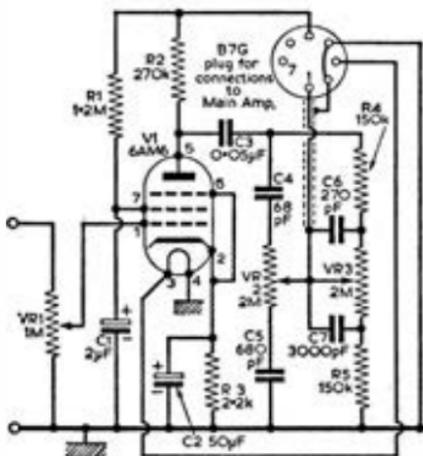
resistor in this stage must be bypassed, not to avoid degeneration but to remove hum caused by the heater-cathode capacitance of the valve. The volume control VR1 is incorporated in the grid circuit. No decoupling is necessary between V1 and V2.

If the expense of an additional 1MΩ potentiometer is no objection, one may be fitted in the grid circuit of V2 as the main volume control in place of R6, so that any noise voltages originating in the first stage will be reduced along with the signal. It will be necessary, however, to retain the control in the grid circuit of V1 to prevent overloading with high level inputs, but a cheaper pre-set type is suitable for this.

Power Supply

The mains transformer must have an output of 350V at 60mA and 6·3V at 2A. The transformer must be securely clamped to prevent any audible buzz from the stampings. Ex-government transformers, impregnated for tropical use, are excellent and one was used in the prototype.

A 6X5 rectifier is shown in Fig. 25, but any type to suit the transformer will be satisfactory. Mains smoothing is provided by R18 in conjunction with C13 and C14; the resistor value may need slight adjustment to produce the right voltages at the valve electrodes. Generous smoothing almost completely eliminates hum and it is very difficult indeed to discover by ear whether the amplifier is switched on.

Construction

The construction is quite straightforward and the layout is in no way critical. The prototype was built in two sections so that the main amplifier could, for physical stability, be fitted at the bottom of a floor-standing cabinet (to be described), and the pre-amplifier and controls higher up in

a convenient operating position. A plan of the main chassis, which is of 16 s.w.g. aluminium sheet, is given in Fig. 27.

The power supplies for the pre-amplifier are taken from the B7G socket on the main amplifier and as the gain from this point onwards is not high, the signal from the pre-amplifier can be passed into the same socket in a screened cable without fear of hum pick-up.

If it is more convenient, there is no reason why the pre-amplifier should not be built on the main chassis and in this
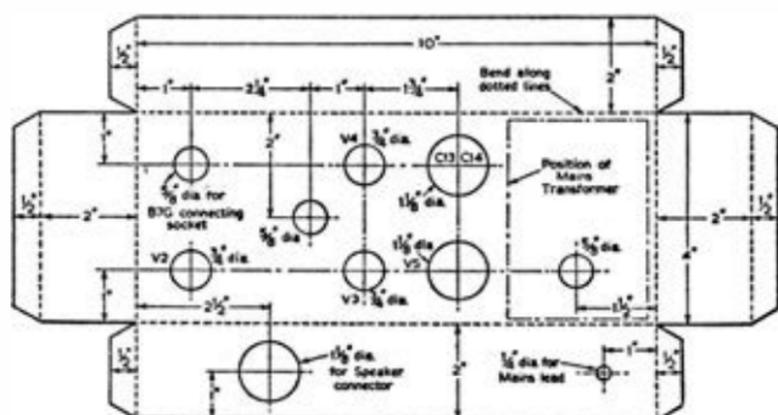
Fig. 27. Dimensions for the main amplifier chassis.

case, the B7G socket will do for V1. The volume and tone controls can be fitted along the front runner, the chassis being made about $1\frac{1}{2}$ in. wider to accommodate them. No other alteration to the layout is necessary.

Pre-amp Construction

A plan of the separate chassis on which the pre-amplifier is built together with the controls, is shown in Fig. 28. It is also of 16 s.w.g. aluminium and is designed to be screwed to the inside of the cabinet behind the control panel. The valve is mounted on an inch wide bracket fitted

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across the chassis at a point $2\frac{3}{4}$ in. from one end—see diagram.

The valve and its immediately associated components can conveniently be made up as a sub-assembly and fitted to the chassis with self-tapping screws after the controls have been fitted and wired. The control panel carries also the mains
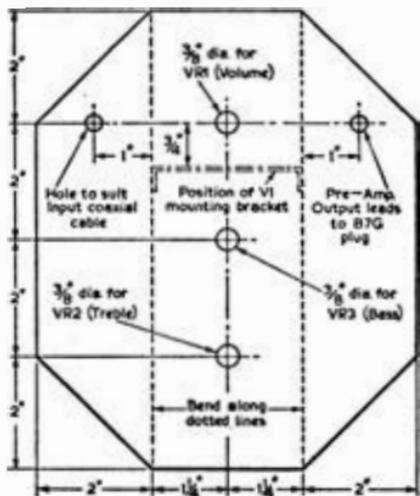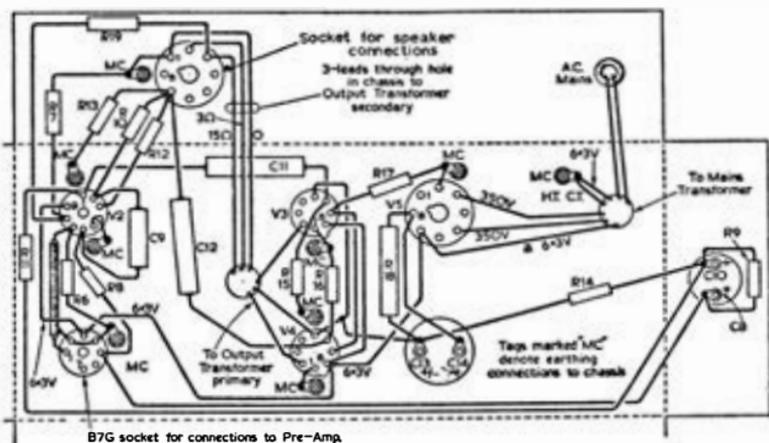

Fig. 28 (left). *Details of the pre-amplifier chassis and support bracket for VI.*
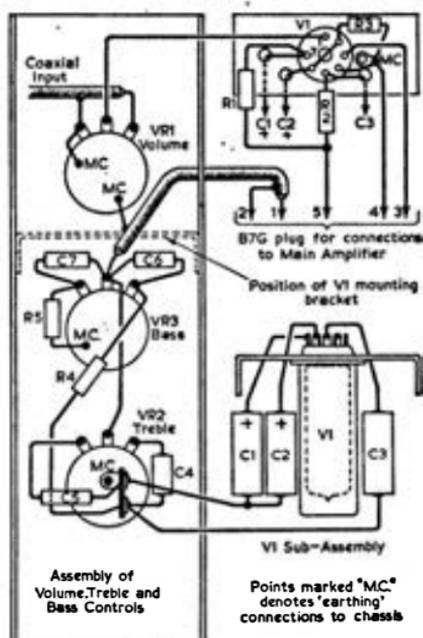
Fig. 29 (below). *Main amplifier wiring diagram.*

Fig. 30. Wiring diagram of the pre-amplifier, with V1 shown out of position for clarity.

switch and pilot light for all the equipment in the cabinet.

If single chassis construction is adopted, it is recommended that the mains switch should be either a separate control or incorporated with VR2, where it is not likely to introduce hum.

The prototype was required only for radio reproduction but if other inputs are to be used regularly, it is suggested that the control panel be enlarged to accommodate a coaxial socket and changeover switch, the wiring for which should be thoroughly screened.

Wiring

The wiring diagrams are given in Figs. 29 and 30. It will be seen that the double electrolytic C13, 14 is mounted in the vacant space beneath the mains transformer. If a drop-through type of transformer is used, it must be fitted

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elsewhere, but there are several alternative positions and the choice is entirely one of convenience.

Tinned copper wire of 22 s.w.g. is suitable for wiring up, and lengths of more than an inch or so should be covered with sleeving. An octal socket is used for connecting the loudspeaker (Fig. 29) the spare tags being used for anchoring one or two small components.

Components

All the resistors may be ½W except R18 (10W) and R17

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(1W). The two 47kΩ load resistors in the invertor stage and the grid resistors in the output stage must be matched fairly accurately and if this cannot be done, 5% components or better should be used.

It is advisable that all the capacitors except C2, should be 500V, so that in an emergency they will be able to withstand the peak output from the rectifier.

Of the valves, V1 may be almost any voltage amplifying pentode and V2 any double triode, though a low μ type is to be preferred.

Testing

When construction is complete, make a check with a meter between the chassis and the rectifier cathode (pin 8) to see that there are no shorts in the h.t. circuits. If all is well, the power can be connected and a measurement made of the voltage between C13 and chassis. It should be about 265V which, allowing for bias, gives about 250V between the anodes of the output valves and their cathodes.

Instability

If there is instability or the reproduction sounds distorted, it will probably be caused by incorrect connection of the feedback circuit. Either the primary or secondary connections to the output transformer should be reversed.

Installation

Lack of space often precludes any serious attempt at a high fidelity speaker enclosure and so it is easier to rely on a simple standing box-baffle, on top of which a tuner can stand in a smaller cabinet. The baffle shown is suitable for an 8 in. or 10 in. speaker and houses, in addition the amplifier, the pre-amplifier and control panel.

Making a Cabinet

The dimensions of the cabinet are given in Fig. 31. Suitable material is plywood not less than $\frac{1}{2}$ in. thick and much time and hard labour will be saved if the supplier can be persuaded to cut the pieces accurately to size.

Assembly is by simple butt joints, glued and screwed with $1\frac{1}{4}$ in. wood screws. These should be inserted about three or four inches apart, along each joint, in holes drilled previously to receive them. The holes should be countersunk so that the screw heads finish up just below the surface.

Fig. 31. A suitable cabinet for the amplifier.
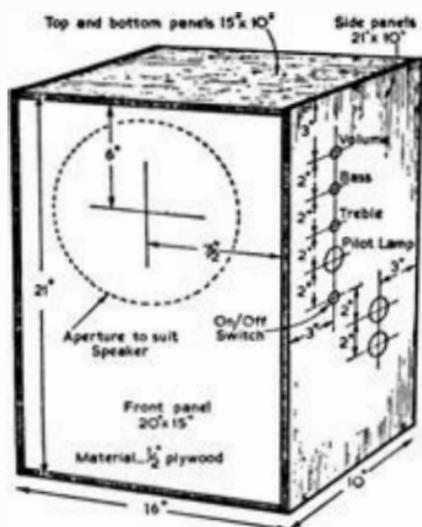
The first operation is to cut a hole in the front for the speaker. This is best done by making a 1 in. hole with a brace and bit on the circumference and cutting away the unwanted material with a coarse hack saw blade in a pad handle. The aperture need not be trimmed off—a rough sanding to remove splinters is enough.

Assembly

The top, bottom and sides should be assembled around the front for trial, using a couple of screws at each corner; the

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front should be a good push fit into the rectangle thus formed.

When all is well, dismantle and make holes in the right hand side for the controls as shown in Fig. 31, after which the top, bottom and sides can be assembled permanently with glue and screws. Push the front into position and set aside for the glue to harden. Do not secure the front yet.

When the glue is hard, remove the front and cover it with a piece of speaker material. This is best secured to the wood with an impact adhesive at the edges. Stick one edge first and allow it to dry thoroughly; the material can then be stretched tightly over the board and the remaining edges secured, after which the front can be fitted permanently to the cabinet.

Apply glue to the mating surfaces and push it into position. Place the assembly face down on a flat surface and adjust the position of the front so that the outer surface of the speaker material is flush with the sides of the cabinet; secure with screws.

Finishing

The outer surface of the cabinet must be sanded off to remove any projections and covered with some sort of material which can be stained and polished. Veneer should be avoided unless the constructor has experience in its application but the same result can be achieved with 2mm hardwood ply, having an oak, walnut, etc., facing. This should be cut to size and secured with an impact adhesive to the outside of the cabinet. Care is necessary to ensure that the surfaces are brought together in exactly the right position at the first attempt.

Cut first a sheet for the front, $15\frac{1}{4}$ in. $\times$ $20\frac{1}{2}$ in. and make an aperture of the desired shape and size. If this is a rectangle as in the prototype, the cut-out portion can be used for the top of the cabinet.

Apply the remaining "frame" to the cabinet in such posi-

tion that it is flush at the top, projects $\frac{1}{8}$ in. each side and leaves $\frac{1}{2}$ in. of uncovered cabinet at the bottom. Cut and fix plywood to the sides in the same way, remembering to make holes in the right-hand side for the controls. The ply for the top should measure 15 in. × 10 in. so as to fit flush at the rear and leave a 2mm rebate at the front and sides when the job is finished.

The outer ply surface must now be prepared for final polishing.

When polishing is done, the $\frac{1}{2}$ in. uncovered strip round the bottom can be enamelled black and the appearance can be further enhanced by adding to the bottom two wood battens about $7\frac{1}{2}$ in. × 1 in. × $\frac{1}{2}$ in. as "feet" so that the cabinet will stand $\frac{1}{2}$ in. above floor level.

Fig. 32. Layout of a control panel.

Control Panel

The control panel may be of tinted Perspex, etc., to blend with the cabinet, or a contrast can be produced by using black or white material. Laminated plastic sheet as used for covering kitchen surfaces is very suitable but must be cut and drilled rather carefully to avoid chipping the polished surface. Cut to the dimensions given in Fig. 32 and secure to the cabinet with chromium plated screws.
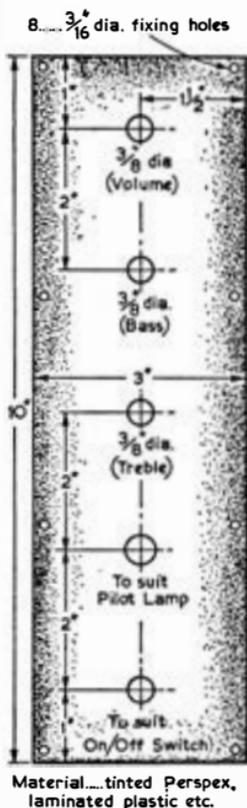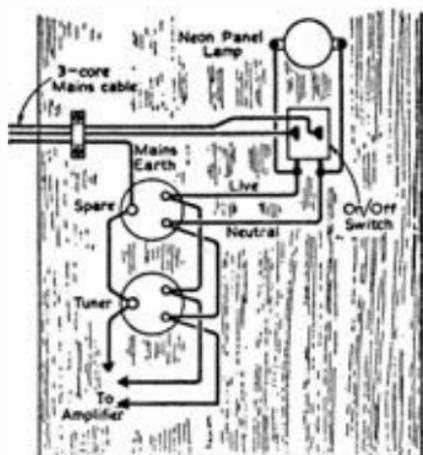
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Fig. 33. Mains connections which are made to the switch, sockets and lamp inside the cabinet.

Loudspeaker

The performance of the amplifier merits a good quality speaker. The prototype used an 8 in. cambric cone type having a wide frequency response.

The speaker should be mounted first upon a piece of $\frac{1}{2}$ in. or $\frac{3}{4}$ in. fibre board (about 10 in. square for an 8 in. speaker), with a suitable cut-out. This sub-baffle should then be secured to the inside of the cabinet over the aperture provided.

To avoid resonances, it is advisable to line the inside of the cabinet with sound absorbent material; ordinary carpet felt secured with adhesive is satisfactory.

Connections

The mains supply to the cabinet should be taken first to the switch and then distributed to sockets mounted inside the cabinet in the position shown in Fig. 31. The tuner can be plugged into one of these sockets. The second is useful for additional apparatus. The wiring is shown in Fig. 33.

The amplifier can be secured to the floor of the cabinet in any desired position by wood screws through the chassis flanges.

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Five-valve Stereo
Amplifier

THIS chapter, it is hoped, will firstly give guidance to those about to enter the field of stereophonic sound, and secondly enable mono hi-fi enthusiasts to convert their present systems to stereo by adding another amplifier.

The Complete System

A complete stereo system is shown diagramatically in Fig. 34. Radio has been included, although stereophonic radio is still only being broadcast on an experimental basis.

Stereo recordings give us two separate sets of intelligence on the record, and these are kept separate throughout the
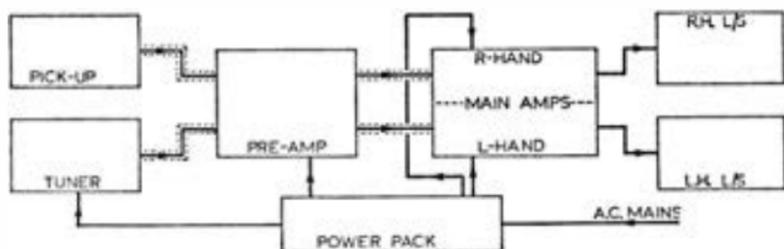

Fig. 34. A complete system for stereo reproduction.

two amplifier and loudspeaker systems. Our ears recombine these two sets of information and give the illusion of depth and breadth of music.

A study of Fig. 34, shows that it comprises a turntable and pick-up, the pre-amplifier and tone control unit, the main amplifiers (preferably identical), the two loudspeaker systems, a radio feeder unit (if fitted), and the power supply.

The Turntable

If the reader already possesses a turntable, it is advisable to write to the manufacturer to ensure that the unit is suitable for stereo. Some of the earlier 3-speed units are not suitable for conversion. If, on the other hand, a new unit is to be purchased, there is a large choice.

Transcription Motor

If expense is no object, a transcription type motor is undoubtedly the best, but for the majority of constructors, a good quality autochanger or single player will be more than adequate and the money saved can be put to good use on the loudspeaker system.

Autochanger

Since their first appearance many years ago, fierce arguments have taken place over the merits and otherwise of autochangers. However, modern changers are vastly superior to their earlier counterparts, and if the reader likes to play a sequence of records, an autochanger is the obvious choice.

Before deciding upon a unit, the following points should be borne in mind: it should be a four speed (78, 45, 33 and 16 rev/min.) unit; it should be possible to adjust the playing weight of the pick-up to between, say 4 and 8 gm.; both the stylus and the cartridge should be easy to change; there must be sufficient clearance both above and below the

motor board for the unit chosen (6 in. above and 2 in. below is a typical value).

Before finally deciding, hear each unit working and listen for the following three points.

Wow. This consists of slow changes in the speed of rotation of the turntable and may best be heard on a long sustained sound, or on piano music, the ear being very sensitive to slight changes of pitch on a continuous note.

Flutter. This is high frequency modulation superimposed on the music, resulting in a "roughness" to the tone of the recording. It is usually less noticeable than "wow".

Rumble. Rumble is of a very low frequency—hence the name—and is best heard on a record with no, or only light modulation, such as a light violin passage. An electrical filter is given later, which can be used to minimise the effect.

The Pick-up

If an autochanger is purchased, a pick-up will normally be fitted. This will almost invariably be of the crystal type, and have advantages of a good frequency response (40-12,000 c/s is quite common); a good output (100 to 300mV from each channel on stereo); and be inexpensive and quite robust.

However, it must be remembered that crystal pick-ups are high impedance devices and a few elementary precautions can bring excellent results.

First, they should be fed into an impedance of $1M\Omega$ if possible, but certainly not less than $0.25M\Omega$, or a falling off of the bass response will result.

Second, the shorter the leads from the pick-up to the pre-amplifier the better, and a maximum of two feet should be allowed. This lead can represent quite a high capacity in shunt with the pick-up and can result in heavy losses. A screened lead should be used to prevent hum pick-up, but

the screening should be insulated and earthed only at the pre-amplifier input. If hum is objectionable, the main power transformer should be re-orientated to find the optimum position.

As a general rule, a crystal pick-up can be represented as a capacity of the order of $0.01\mu F$ in shunt with several megohms. For this reason, shunting of the pick-up output by a capacitor does not result in a high frequency cut, as in the case of a magnetic pick-up, but only in a reduction of the output as a whole. Also, in some cases, a correction circuit is recommended by the pick-up manufacturer and this usually consists of a simple capacity/resistance circuit.

Before leaving the subject of turntables and pick-ups it is worth mentioning a few of the faults of stereo cartridges.

They are more responsive to turntable rumble, owing to the fact that they must respond to a vertical as well as horizontal component of the recording. Acoustic feedback through the cabinet can be a serious problem, unless the baseboard and cabinet are very substantial, and the speakers well shielded from the turntable. Unless there is a definite reason against it, the loudspeakers should be separately housed.

The Radio Feeder Unit

The only decisive factor to bear in mind here is whether to use f.m. or a.m. for reception; f.m. is the obvious choice, and unless there are any other strong reasons, a.m. may be ignored.

The h.t. and l.t. requirements are dealt with under the power-pack section, but in general, most units require an h.t. between 200 and 250V and take 30 to 50mA.

Pre-amplifier and Tone Control Unit

The function of this single valve unit is to provide some measure of gain, to provide switching for stereo records,
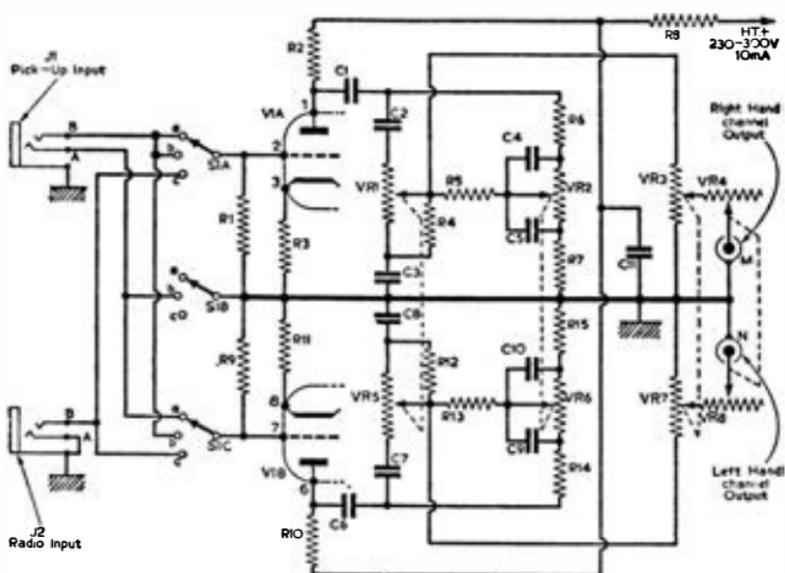
Fig. 35. Pre-amplifier and tone control circuit. Note: when VR4 is turned clockwise the resistance must decrease; VR8 must operate in the opposite sense. Cables linking main and pre-amplifiers should be single-cored, flexible and screened.

mono records and radio (another position can be included if tape is to be used), and tone, volume and balance controls.

In this particular design, this unit has been kept as a separate chassis since all the amplifier controls are on this unit and can, therefore, be bolted to the control panel of the radiogram, leaving the heavier units such as power-pack and main amplifiers to be housed at the bottom of the cabinet. This arrangement is also less likely to suffer from hum pick-up, as the pre-amplifier can be kept close to the pick-up and well away from the mains transformer.

If we now consider the detailed circuit (Fig. 35) we see that a single double triode valve (12AT7) is used. This is purely for economy purposes, and two EF86s connected as triodes would be a slightly better arrangement. One half of the valve is used as a pre-amplifier for the right hand chan-

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nel and the other half for the left hand channel, each circuit being identical.

Taking one half of the circuit, it can be seen that the pick-up is fed into a 2-pole jack, J1, on to the "selector switch" S1. This selector switch arranges that in position "a" the common lead of the pick-up is earthed, and the left and right hand channel leads are connected to their respective grids.

With the cathode resistors R3 and R11 not bypassed, a gain of 26 is obtained (if bypassed by a 50μF 12V capacitor, a gain of 40 is obtained), and the amplified signal is fed via C1 to the tone control circuits.

Network C2, C3 and VR1 forms the treble boost and cut control and R6, VR2, R7, C4 and C5 constitute the bass boost and cut circuits. VR3 is the master volume control, and VR4 the balance control. The tone control circuits introduce a loss of approximately 10 to 1, so that the overall gain of the pre-amplifier is of the order of two. This is adequate since the main amplifiers need only 100mV for full ouput. The overall distortion of the pre-amplifier circuit is less than 0·2%.

Let us now consider each control and its function in a little more detail.

Selector Switch (SI)

As has been said already this has three positions, stereo (a), mono (b) and radio (c), and a forth (tape) can be added if needed.

Treble

VR1, VR5, is a twin-gang, linear, 250kΩ potentiometer and provides a boost of approximately 10dB and a cut of 10dB, continuously variable (measured at 10kc/s relative to 1kc/s).

Bass

VR2, VR6 is a twin-gang, linear, 250kΩ potentiometer and provides a boost of approximately 10dB and a cut of 5dB continuously variable (measured at 50c/s relative to 1,000c/s).

COMPONENTS LIST FOR PRE-AMPLIFIER

Resistors:

R1 1MΩ 5% H.S.	R5 39kΩ	R9 1MΩ 5% H.S.	R13 39kΩ
R2 100kΩ	R6 68kΩ	R10 100kΩ	R14 68kΩ
5% H.S.		5% H.S.	R15 6·8kΩ
R3 1,500Ω	R7 6·8kΩ	R11 1,500Ω	
R4 47kΩ	R8 15kΩ	R12 47kΩ	

All ±10% ½W carbon, unless otherwise stated

Potentiometers:
VR1, VR5 250kΩ + 250kΩ, lin.
VR2, VR6 250kΩ + 250kΩ, lin.
VR3, VR7 250kΩ + 250kΩ, log.
VR4, VR8 500kΩ + 500kΩ, lin.
All dual ganged.

Capacitors:

C1 0·25μF paper 350V	C7 600pF silver mica
C2 600pF silver mica	C8 8,200pF silver mica
C3 8,200pF silver mica	C9 300pF silver mica
C4 300pF silver mica	C10 3,300pF silver mica
C5 3,300pF silver mica	C11 0·25μF paper 350V
C6 0·25μF paper 350V	

Miscellaneous:
V1 ECC81 or 12AT7
J1, 2 2-pole earth input jack socket
S1 3-pole, 3-way rotary switch
B9A valveholder, nylon loaded, skirted and resiliently mounted. About 3 ft. of light, twin-core screened cable (for wiring pick-up to input socket). Suitable length of single-core screened cable to connect output of each channel to main amplifier.

Master Volume Control

VR3, VR7 is a twin-gang, logarithmic law, potentiometer, providing simultaneous volume control of each channel. A log. law potentiometer is needed here in order to avoid the "fierceness" at the minimum volume position which would result if a linear potentiometer were used.

Balance Control

VR4, VR8 is a twin-gang 500kΩ linear potentiometer, the function of which is to adjust the volume from the two loudspeakers for equal level. In theory, if everything in the amplifiers and loudspeakers were identical, this control would not be needed, but owing to tolerances in components, and room acoustics, this control will be found very useful.

VR4 is wired up so that its operation in a clockwise direction decreases the resistance; VR8 is wired up in the opposite sense.

The cables linking the pre-amplifier to the main amplifier should be single-core, flexible, screened cable.

Reverse Stereo

A reverse stereo control has not been fitted. The function of such a control is to reverse the output to the speakers, i.e. to change the right hand channel to the left hand loudspeaker and vice versa. If such a control is considered necessary, a simple change-over switch can be fitted to change over the outputs M and N (Fig. 35) in such a way that, in one position, M can be fed into the right hand channel main amplifier and by reversing the switch into the left hand main amplifier. N is simultaneously changed over in the opposite sense.

Precautions in Wiring

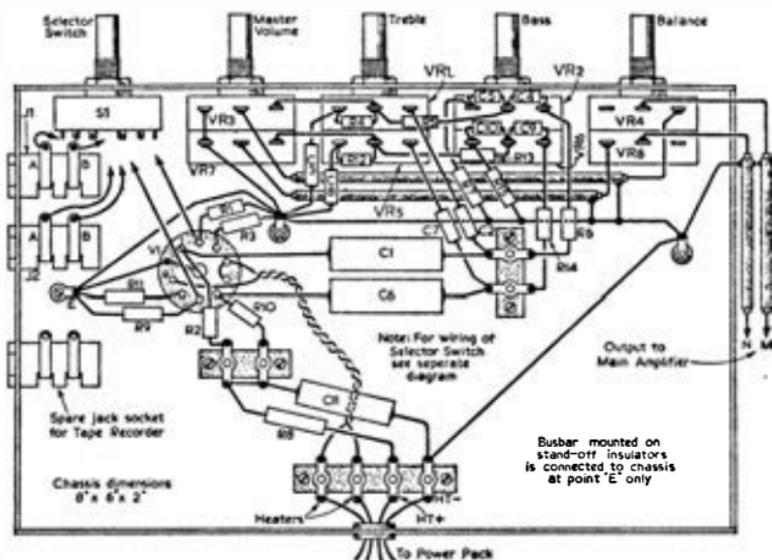The following points should be considered standard practice in any high quality sound reproducing equipment
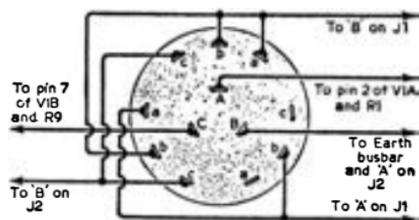
Fig. 36a. Pre-amplifier wiring diagram. J1, J2 and the spare socket should be positioned as close to S1 as possible.

Fig. 36b. Wiring details of the selector switch, S1.

and if observed in the construction of the pre-amplifier (see Fig. 36) hum and noise will be inaudible.

1. High stability resistors should be used where specified, otherwise carbon composition types are quite suitable.

2. The valveholder should be of the loaded nylon, resilient type, and if the loudspeakers are housed in the same cabinet, acoustic shielding in the form of a wooden partition should be arranged between the loudspeakers and the pre-amplifier.

3. The mains power transformer should be kept as far away from the pre-amplifier unit as possible, and if hum is found, re-orientation of the transformer may cure the trouble.

4. A good earth bus-bar of 12 or 14 s.w.g. tinned copper wire should be run on insulators around the chassis and earthed to the chassis at one point only, the input jack, J1. All earth connections should be made to this bus-bar and not to the chassis.

5. Heater leads should be twisted closely together, kept close to the chassis and well away from input and grid leads.

J1
Tag A is connected to Left Hand channel grid via S1C
Tag B is connected to Right Hand channel grid via S1A

J2
Tag A is connected to earth
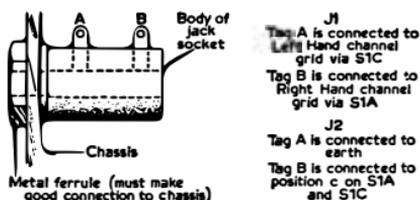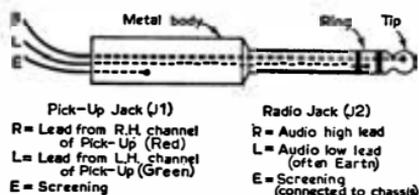Tag B is connected to position c on S1A and S1C

Fig. 37. Connections details of J1 and J2.

6. Input leads to the grid circuits should be kept as short as possible, and if the above instructions are carried out, no screened leads need be used in the pre-amplifier and internal shielding will not be required.

When considering the power output per channel, it will be found that stereo has the effect of apparently louder volume for a given power output. In the largest room likely

Fig. 38. Connections to the input jack plugs.

Pick-Up Jack (J1)
R = Lead from R.H. channel of Pick-Up (Red)
L = Lead from L.H. channel of Pick-Up (Green)
E = Screening

Radio Jack (J2)
R = Audio high lead
L = Audio low lead (often Earth)
E = Screening (connected to chassis)

to be found in a normal house, 3W per channel will be more than adequate.

To keep cost and hence the valve complement to a minimum, single ended amplifiers are essential, and since we need only 3W per channel, by using heavy negative feedback, distortion can be kept to a minimum.

Fortunately a design already exists in the Mullard 3-3 amplifier, which is very small, inexpensive to build and

excellent in operation. A simplified version of the Mullard 3-3 is used and a full circuit diagram of one amplifier is shown in Fig. 39. The Mullard publication "Mullard Circuits for Audio Amplifiers" will be found helpful.

The Circuit

Considering only one channel (the right hand channel), the output from the pre-amplifier is fed to the main amplifier by a single-core screened cable, terminated in a single-pole jack plug. This jack plug is inserted into a single-pole jack socket (J1) on the main amplifier chassis.

The first valve (V1) is an EF86, a low noise, low microphony, pentode (somewhat similar in its electrical character-
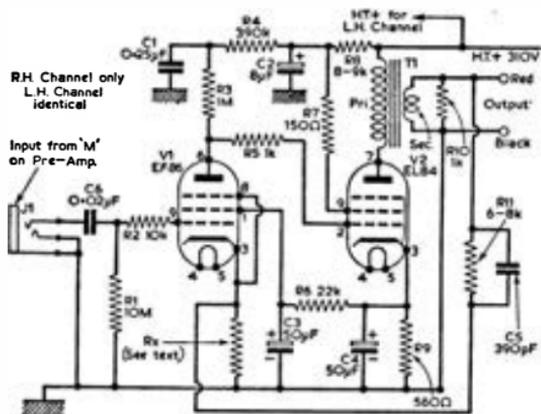

Fig. 39. One of the main amplifiers—this circuit is duplicated for the other.

istics to the older 6J7). The EF86 is used under starvation conditions, with anode and screen voltages much lower than is usual. The total current consumption is low and a $1M\Omega$ high stability resistor is used as an anode load. It is important to use a high stability resistor here to prevent noise.

The circuit configuration results in very high gain, and the valve is direct-coupled to a high slope output valve, an EL84. Such high gain is achieved that approximately 26dB of feedback can be used, and this is taken from the second-

COMPONENTS LIST FOR ONE MAIN AMPLIFIER

Resistors:

R1 10MΩ	R4 390kΩ	R7 150Ω	R10 1kΩ
R2 10kΩ	R5 1kΩ	R8 3·9kΩ	R11 6·8kΩ
R3 1MΩ	R6 22kΩ	R9 560Ω	Rx 120Ω for 3Ω;
5% H.S.		3W w.w.	82Ω for 15Ω
			speaker

All ±10% ½W carbon, unless otherwise stated

Capacitors:

C1 0·25μF paper 350V   C4 50μF electrolytic 50V
C2 8μF electrolytic 450V   C5 390pF silver mica
C3 50μF electrolytic 50V   C6 0·02μF paper 350V

Miscellaneous:

T1 Output transformer: Parmeko P2641, Colne 03077, Wynall W1452, Gilson W.O.767 or Partridge P4073
V1 EF86   V2 EL84
J1 2-pole input jack socket
One B9A valveholder, nylon loaded, skirted. One B9A valveholder, moulded type. Loudspeaker output socket. Aluminium chassis 8 in. × 6 in. × 2 in. (this accommodates both main amplifiers).

This components list should be duplicated to cover the second main amplifier.

ary of the output transformer and fed back into the cathode of the EF86. The overall result is 3W output at less than 1% total distortion, and a good damping factor for the loudspeaker.

It should be noted that the anode of the EF86 is directly coupled to the grid of the EL84 via R5 and the screen voltage for the EF86 is obtained from the cathode of the EL84. In this way a high degree of stability of electrode voltage is obtained.

Feedback is supplied by R11 and C5 in shunt and no

variable controls are needed on the modified chassis, since these are taken care of in the pre-amplifier and tone control unit.

The secondary of the output transformer is taken to colour-coded sockets on the chassis, and a 1kΩ resistor is shunted across the output. This is to prevent instability and possible damage to the output transformer if the loud-speaker should be inadvertently disconnected.

The same wiring precautions should be adopted as detailed under the pre-amplifier and tone control unit.

Both main amplifiers may be built on a single 8 in. × 6 in. × 2 in. chassis (Fig. 40) and it is advisable to mount the output transformers at 90° to each other to minimise coupling.

Output Transformer

The components list should be closely followed, and the only component which needs special mention is the output transformer. This is quite straightforward and has an 8H primary inductance and secondary to match 3 or 15Ω speakers. It should be noted that resistor Rx should be altered depending upon which output impedance is used (see components list and Fig. 39).

The Loudspeaker System

Several full length chapters could be written on this sub-ject alone, and most people have their own idea of what is the best arrangement. It would be, however, useless to build an amplifier with low distortion and then use ordinary, small-size speakers built into a small radiogram cabinet. The choice of speaker will depend upon space and finance available and it is suggested that one of two systems be used.

With a small reflex enclosure, a good single full range high fidelity loudspeaker is suitable. A frequency range of 40 to 15,000c/s, 6W handling capacity and a fundamental resonance of 65c/s is typical of such a speaker.

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With a larger reflex enclosure, starting perhaps with one loudspeaker, provision for the addition later on of a mid-range and tweeter speaker is advisable.

Placing of the Loudspeakers

As a general rule the aim should be to keep the loud-speakers separate from the radiogram itself for the follow-

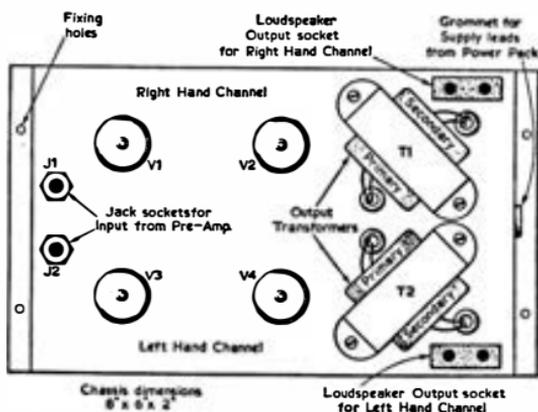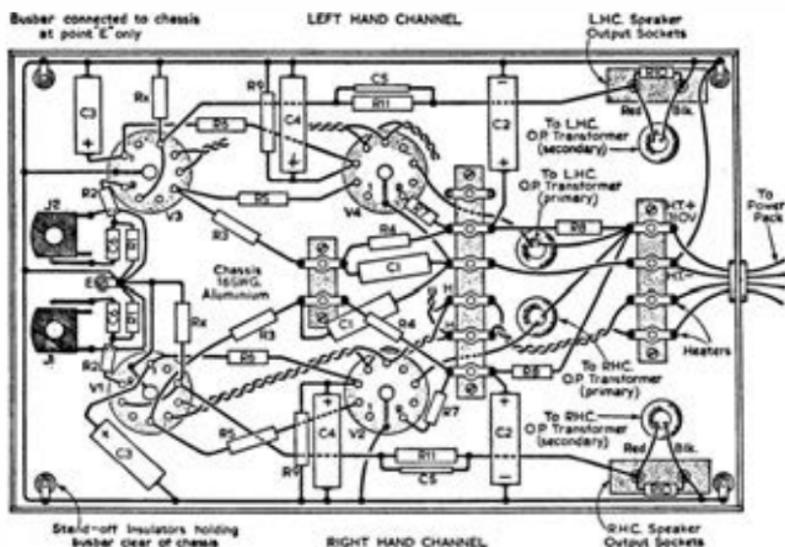Fig. 40. Above-chassis layout for the twin main amplifiers.

Fig. 41 (below). The underchassis wiring diagram.

Note: the duplicated resistor and capacitor numbers represent the twin circuits for the main amplifier. Ensure correct polarity of electrolytics.

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ing reasons: (a) acoustic feedback can more easily be eliminated; (b) very large and solidly built radiogram cabinets for good bass response are not usual; (c) positioning of the loudspeakers for best results is made possible.

Each room is different acoustically and in order to find the best position for maximum stereo effect, some experimenting will be necessary. As a guide, begin with the arrangement shown in Fig. 42 (a); then try Fig. 42 (b). Other positions
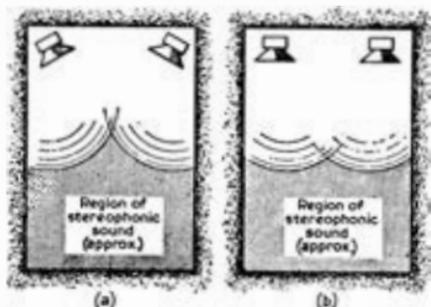

Fig. 42. Regions of stereophonic effect obtained in a small room with different speaker positions.

can be tried, but the loudspeakers should always be kept between 4 and 8 ft. apart.

The Power Supply

Most enthusiasts will have suitable components for the power supply available, and provided that the h.t. voltages given are fed to the various units, satisfactory operation will
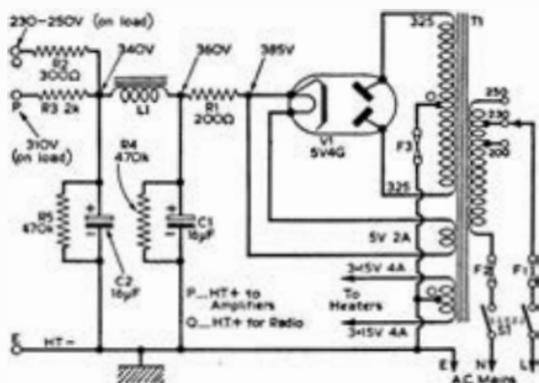

Fig. 43. Power supply circuit.

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be ensured. It is essential that the h.t. fed to the units has some degree of smoothing such as choke and capacitor, and that the 6·3V heater supply is centre-tapped to earth.

However, for those wishing to build a new power supply, a suitable design is given in Fig. 43. Remember, always feed the unit from a 3-pin plug/socket arrangement and ensure that earthing is efficient.

COMPONENTS LIST FOR POWER PACK

Resistors:
R1 200Ω 10W w.w. R3 2,000Ω 10W w.w. R5 500kΩ 2W
R2 300Ω 6W w.w. R4 500kΩ 2W
NOTE. R2 and R3 can be adjusted on load to provide correct h.t.

Capacitors:
C1 16μF electrolytic 500V
C2 16μF electrolytic 500V

Miscellaneous:
F1, F2 2A fuse
F3   250mA fuse
L1   Smoothing choke 10H 150mA
S1   D.P.S.T. toggle switch
T1   Mains transformer. Secondaries: 325-0-325V 150mA: 5V 2A 6·3V (centre tapped) 4A
V1   5V4G

A few fundamental precautions are worth listing.

(1) All electrolytic capacitors should be adequately rated, e.g. a 325-0-325 secondary will produce a peak voltage on load of 450V, and without a load may well be over 500V.

(2) An indirectly heated rectifier is better for capacitors than a directly heated rectifier.

(3) Make sure the chassis is earthed.

(4) Fit a fuse in the leads of the mains input and another in the lead from the centre-tap of the secondary h.t. winding to earth.

(5) Bleed resistors of 470kΩ should be fitted across each electrolytic capacitor in the h.t. smoothing circuit.

(6) Keep the power supply well away from the pick-up and pre-amplifier unit.

(7) If possible, use a separate heater winding for the rectifier valve.

Power Supply Requirements

Pre-amplifier and tone control unit:

   h.t. = 230 to 250V at 10mA

   l.t. = 6·3V at 0·4A

Main amplifiers:

   h.t. = 310V at 95mA

   l.t. = 6·3V at 2·0A

Radio feeder:

   h.t. = 230 to 280V at 40mA

   l.t. = 6·3V at 2·0A (to allow for dial lamps).

The total h.t. current consumption can be taken to be

TABLE I

MAIN AMPLIFIER TEST VOLTAGES

(using high resistance voltmeter)

H.T. 310V at 48mA		
at C2	285V	
at C1	210V	
V2 anode voltage	290	
screen voltage	285	(approximate)
cathode voltage	28	
V1 anode voltage	20	
screen voltage	28	

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about 150mA, and the l.t. as 6·3V at 4A together with a winding to suit the rectifier; e.g. 5V 2A, for a 5V4G rectifier.

Setting up the Completed Unit

Having checked the wiring, a few preliminary adjustments are necessary.

(a) Pre-amplifier.

Apart from h.t. and l.t. voltage, the only precaution is to ensure that the right hand channel lead from the pick-up goes to its appropriate grid input and that the audio output from the pre-amplifier is connected into the right hand channel main amplifier. The same check should be made on the left hand channel.

(b) Main Amplifiers.

Firstly, to obtain correct phasing of the feedback connect the main amplifiers, but first disconnect the feedback lead in, say, the left hand channel. If possible use an old loud-speaker and connect it across the loudspeaker output leads of the right hand channel. Load the loudspeaker output of the left hand channel with a 3Ω 1W resistor. Switch on h.t. and l.t., and if violent oscillation occurs, switch off. Reverse the secondary connections to the output transformer of the right hand channel, when all should be well. Switch off, remove the loudspeaker from the right hand channel output sockets to the left hand output sockets and load the right hand output sockets with the resistor. Reconnect the feedback in the left hand channel. If violent oscillations occur on switching on, reverse the secondary connections to the left hand output transformer, when again all should be well.

Secondly, switch off and connect up the correct loud-speakers to their respective channels.

(c) Loudspeaker phasing.

It is essential that the loudspeakers are correctly phased, and this can be arranged as follows.

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If each loudspeaker system consists of more than one loudspeaker, each speaker in that system must be correctly phased. Take each loudspeaker separately and apply a 1½V battery across the speech coil, and note which way the cone moves on connecting the battery. It is worth using a convention here, and the following is suggested.

If the cone moves inwards, reverse the connection to the battery so that the cone now moves outwards when the battery is connected. Mark the speech coil solder tag connected to the positive of the battery with a +. Repeat this for each loudspeaker in the system.

Correct phasing will be obtained when all + signs are connected to same lead. Also, code the lead connecting these + signs with a red sleeve, so that it can be connected

Fig. 44. Left—*Original pre-amplifier circuitry*; right—*modified circuitry to remove "rumble".*
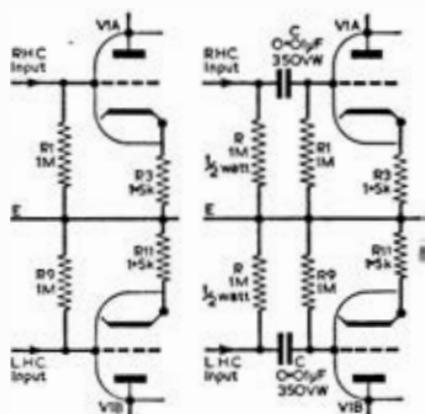
to the loudspeaker output sockets on the main amplifier in the correct way.

Repeat the above for the other channel loudspeaker system.

Rumble Filter

If turntable rumble is pronounced or noticeable when playing quiet passages, the input circuit of the pre-amplifier may be modified as shown in Fig. 44.

Design for a
Bass Reflex Cabinet

ONE of the most difficult problems encountered in the design of "hi-fi" equipment is that of obtaining a level and undistorted bass response. For this reason the design of the loudspeaker cabinet should be undertaken most carefully.

Unmounted Loudspeaker

The vibrating cone of a moving coil loudspeaker is small compared with the wavelength of low frequency sounds in air. If a loudspeaker is used at low frequencies without any baffle whatsoever, the air which is pushed forward by the loudspeaker cone will merely pass around the outside of the loudspeaker to fill the partial vacuum created at the rear of the cone. Similarly when the cone moves backwards, air will pass from the rear to the front of the speaker.

The loudspeaker cone is intended to set the whole of the air in the room in vibration, but if no baffle is used, the air slips around the edges of the loudspeaker at low frequencies and the effect on the air in the room is very small indeed.

The air currents (i.e. the sounds) are thus short circuited. The use of a baffle will prevent this short circuiting of the air currents provided that the baffle is not so small that the air can pass easily around the outside of it from the front to the back of the speaker, and vice-versa at the frequency in question.

High Frequencies

At high frequencies, however, the movements of the cone are so rapid that there is not enough time for the air to move between the back and front of the speaker, and so to equalise completely the pressures, before the direction of movement of the loudspeaker cone has been reversed.

One has only to compare the sound from an unmounted speaker with the sound from a properly mounted speaker to appreciate that the unmounted speaker produces almost negligible sound when the input frequency is low. The volume of the middle frequencies is also much less than that from a correctly mounted loudspeaker. This effect is very prominent with a large unmounted speaker, but is even worse with smaller loudspeakers, as the air can pass from the back to the front of the cone so much more easily and quickly.

Loudspeaker Resonance

The resonance curve of a typical loudspeaker of nominal impedance 15Ω is shown in Fig. 45. In this graph the actual impedance of the speaker in ohms is plotted against the input frequency. The graph covers only bass frequencies; the impedance rises somewhat at high frequencies. The peak, in the case shown at 65c/s, is at the natural resonant frequency of the loudspeaker; that is, at the frequency at which the speaker cone vibrates if the cone is gently tapped with the finger. The speaker cone will vibrate with a very
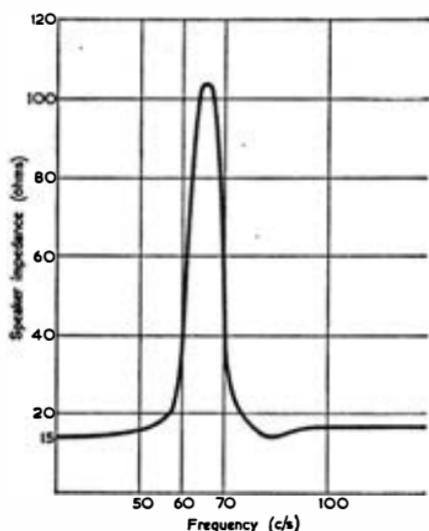
Fig. 45. Here the actual impedance of a loudspeaker of nominal impedance 15Ω has been plotted against input frequency. Only very low frequencies are covered by the graph which shows the speaker's natural resonant frequency to be 65c/s.

large amplitude if quite a small input of this particular frequency is applied to the loudspeaker terminals.

Bass Reflex

A simple system which gives an equally satisfactory performance consists of a plain, closed, cabinet with an opening at the front (Fig. 46): The volume of the cabinet and opening are so calculated that the air inside the cabinet resonates with the air in the "vent" at the same frequency as that of the fundamental resonance of the loudspeaker used. The sound emerging from the vent is in phase with

Fig. 46. Typical cabinet with rectangular vent and tunnel beneath the speaker.

that from the front of the loudspeaker cone. Hence the bass reflex cabinet is also known as the acoustical phase inverter.
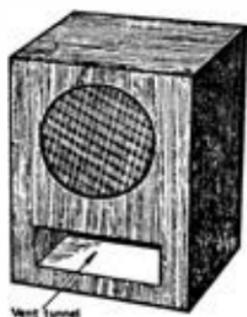
The size of the speaker cabinet required for this arrange-

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ment can be appreciably reduced by fitting a pipe or wooden tunnel which projects back into the cabinet from the opening at the front as shown in Fig. 46.

Design

The first problem one meets when designing a bass reflex cabinet, is that of calculating the size of the enclosure and vent which will enable the cabinet to resonate at the same frequency as the loudspeaker.

In order to do this, one must know the fundamental resonant frequency of the loudspeaker. The manufacturers of all good loudspeakers supply a nominal value for the resonant frequency. However, this frequency usually decreases by about 10% to 15% after the loudspeaker has been in use for some hours. This decrease occurs because the suspension becomes weaker after a few hours of use.

It is therefore best to measure the resonant frequency of a speaker experimentally if possible, not when the loudspeaker is quite new, but when it has been used for about twenty hours or so, fastened to a temporary mounting and used at reasonably high power. The value obtained can then be used in the calculation.

If for any reason the frequency is not measured experimentally, the manufacturer's figure may be used, but it is wise to reduce this figure by up to 10% to allow for the suspension becoming slightly weaker with use.

Experimental Determination

The fundamental resonant frequencey of the speaker may be determined experimentally in the following way. An audio signal generator should be used to feed an

Fig. 47. A set-up for determining the fundamental resonant frequency of a loud-speaker.

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audio power amplifier connected to the unmounted loudspeaker under test. An a.c. ammeter should be placed in one of the leads to the loudspeaker as shown in Fig. 47. The input power from the amplifier to the speaker should not exceed a small fraction of a watt and therefore a reasonably sensitive ammeter should be used, say 0-500mA. The amplitude of the cone movement should be checked (by touching the cone) to make sure that it is not too large.

As the signal generator is tuned and its output kept fairly constant, the current passing through the loudspeaker and indicated by the ammeter will be found to have a definite minimum value at the loudspeaker resonant frequency. This frequency can then be read from the scale of the audio generator.

The Helmholtz Resonator

A bass reflex cabinet resonates in the same way as a Helmholtz resonator. (If one blows across the top of a suitable bottle, a similar resonance effect can be noticed.)

The theory of the Helmholtz resonator has been worked out in detail and it can be shown that the resonant frequency is determined by the volume of the enclosure itself and by the dimensions of the opening or vent. The air in the enclosure acts in a way similar to that of a capacitor in an electrical resonant circuit and the air in the vent resembles an inductance.

In the same way that various combinations of inductance and capacitance can resonate at the same frequency, it is found that Helmholtz resonators or loudspeaker cabinets can be designed to operate at the same frequency and yet have different vent sizes and internal volumes. Once the vent size is chosen, however, there is only one particular volume which the interior of the cabinet can have if it is to resonate at the same frequency as that of the loudspeaker cone.

Formula

According to Rayleigh, the resonant frequency of a Helmholtz resonator is

$$f = \frac{V}{2\pi} \sqrt{\left\{\frac{A}{C(L + \frac{1}{2}\sqrt{[\pi A]})}\right\}}$$

where V = the speed of sound in inches per second.

π = 3·142.

A = Cross sectional area of vent in square inches.

C = Volume of air in the cabinet in cubic inches.

L = Length of vent tunnel in inches (including thickness of front panel).

This equation can be squared all through and transposed to enable us to obtain the cabinet volume.

$$C = \frac{V^2.A}{4\pi^2 f^2 (L + \frac{1}{2}\sqrt{[\pi.A]})}$$

Substituting for the value for the velocity of sound (13,500 inches per sec.) and for π:

$$C = \frac{4,614,000A}{f^2(L + 0.886\sqrt{A})}$$

The Vent

Before the calculation can be continued, the dimensions of the vent must be decided. Opinions differ somewhat as to the most satisfactory values for these. It is generally agreed that the vent should not extend too far into the cabinet, and that the length of the vent tunnel should not be greater than about one twelfth of the wavelength of the resonant frequency of the loudspeaker cone; neither should the length of the vent be so great that it extends more than half way between the front and back of the loudspeaker.

It is not even necessary to make a vent tunnel at all, for if

one merely cuts a hole in the front panel, one will have a vent of a length equal to the thickness of the front panel. If this course is adopted, however, the cabinet will have to be considerably larger than is necessary if it is to resonate at the desired frequency. Wood is expensive and space in a room is scarce, so a vent tunnel is usually employed. It can be seen from the equations that a long vent will enable a smaller cabinet to be used.

The size of the vent hole is also important; i.e. its cross sectional area (A). Some experts advise that the area of the vent should be equal to the area of the effective part of the loudspeaker cone at low frequencies, whilst others think that any reasonable area (say between one half and double the effective loudspeaker cone area) will give satisfactory results. The effective cone area is the area of the sides of the cone (not including the corrugated suspension at the edge) projected on to an imaginary circle at the front of the loudspeaker.

The area of the vent appears in both the numerator and denominator of the above equations, but the term in the numerator is most important; the use of a small vent area will therefore enable a smaller cabinet to be used than the use of a large vent, other things being equal. The vent can be of any shape, but will normally be of rectangular cross section to simplify construction of the tunnel. Circular vents are sometimes used.

Design Example

Let us suppose that we wish to design a bass reflex cabinet for use with a loudspeaker of a nominal diameter of 10 in. which has a fundamental resonant frequency, measured as described, of 65c/s.

It is first necessary to insert values for the length and cross sectional area of the vent in the equation in order to try to design a cabinet of suitable dimensions.

Let us assume that the vent is to be 5 in. long and that the cross sectional area of the vent opening is to be equal to the area of the effective part of the loudspeaker cone. The average 10 in. loudspeaker will have an effective cone diameter of about 8·5 in. and an effective cone radius of half this value, i.e. 4·25 in. The area of the vent will equal πr^2; where r = 4·25 and π = 3·142; the area = 56·75 sq. in.

Calculation

Putting these values into the last equation:

$$C = \frac{4,614,000 \times 56 \cdot 75}{(65)^2 (5 + 0 \cdot 886 \sqrt{[56 \cdot 75]})}$$

$$C = \frac{61,980}{(5 + 6 \cdot 67)} = 5311 \text{ cu. in.}$$

This then, is the volume of air which should be enclosed in the cabinet and does not include the volume of air and wood in the vent. In order to obtain a value for the total volume

TABLE I

Nominal Diameter of Loudspeaker	Approximate volume of air displaced by speaker
6 in.	150 cu. in.
8 in.	250 cu. in.
10 in.	400 cu. in.
12 in.	630 cu. in.
15 in.	1350 cu. in.
18 in.	2400 cu. in.

of the inside of the cabinet, it is necessary to add to the above figure the following quantities.

(1) The volume of the vent (i.e. the external volume of the vent as seen from inside the cabinet including the volume of air and the wood from which the vent is made).

(2) The volume of the loudspeaker. The presence of the loudspeaker reduces the volume of air in the cabinet. If the volume of the loudspeaker is quoted by the makers, this figure should be used. Otherwise the approximate volume of the loudspeaker can be obtained from the data given in Table 1.

(3) The estimated volume of any wooden struts which are to be used to strengthen the cabinet. About 100 cu. in. may be allowed for a fairly small cabinet and more for a large one.

(4) As it is much easier to reduce the volume of the completed cabinet by fastening a block of wood in it than it is to raise the frequency by extending the vent inwards, it is wise to add an additional hundred cubic inches or so which can be kept in reserve. In the design being considered, the volume of air in the vent tunnel is $56 \cdot 75 \times 5$ which is about $283 \cdot 75$ cu. in. if the wood is to be $\frac{3}{4}$ in. thick. About 80 cu. in. may be allowed for the wood of the vent tunnel. From Table 1, the volume of the 10 in. loudspeaker will be approximately 400 cu. in.

Adding these quantities to the 5,311 cu. in. obtained above, we find that the total internal volume of the cabinet should be approximately 6,275 cu. in.

Possible Shapes

A number of different shapes of cabinet can be used to satisfy the condition that the total internal volume must be approximately 6,275 cu. in. The vertical height of the cabinet must be great enough to accommodate the loudspeaker

and vent above one another. It is therefore wise to start by estimating the approximate minimum height. For this purpose let us assume that the vent area of 56·75 sq. in. is to be obtained by making a hole 6 in. in vertical height and 9·46 in. in length in the front of the cabinet. Allow minimum dimensions of:

top of the cabinet to top of the loudspeaker ..	1 in.
distance across loudspeaker, allowing space for fixing	11 in.
distance from the bottom of the loudspeaker to the top of the vent	2 in.
distance across vent	6 in.
distance between bottom of vent and bottom of cabinet	2 in.
Therefore, the total minimum height of the cabinet is	22 in.

The vertical height must therefore be at least 22 in. Similarly the breadth should be at least 12 in. to accommodate the loudspeaker and vent easily. The depth of the cabinet must be at least 10 in. or the requirement that the vent tunnel length must not exceed half the length of the cabinet will not be met.

Three lengths must now be chosen which are possible values for the length, width and height. These values must be compatible with the foregoing requirements and when all three lengths, expressed in inches, are multiplied together the result must be approximately 6,275 cu. in. There are, of course, many possibilities and the constructor must now make his own choice from the possible dimensions. As the tunnel will not extend half way through the cabinet, the calculation could be carried out again with a longer tunnel length to obtain the smallest possible cabinet size.

Other Loudspeakers

The results obtained above apply to a loudspeaker with a resonant frequency of 65c/s. If a loudspeaker with a fundamental resonant frequency of half this value were used, namely one resonating at 32·5c/s, the volumes of air enclosed in the cabinet should be four times as great. Thus the cabinet will be very much larger; it will also require much more support or the use of thicker wood if undesirable vibrations are not going to take place. Thus the cost of the cabinet is increased enormously, but a much better bass response should be obtained.

Construction

Cabinets are normally made of wood, but brick and concrete can be used very successfully. Assuming wood is to be used, it should be quite thick—at least half an inch for the smaller cabinets, but a thickness of an inch is desirable for the larger ones. Cabinets are made in which the walls consist of a double thickness of wood, the space between the inner and outer walls being filled with sand. Such cabinets are extremely heavy but comparatively cheap to build and certainly very effective.

It is desirable to fix adjoining pieces of wood together with glue and also with screws placed three or four inches apart. A hundred or so screws at least will be required. Countersunk screws should be employed, as they can then be fitted with their heads below the surface of the wood. Plastic wood can then be used to cover the heads of the screws. The plastic wood used should be chosen to match the colour of the other wood as closely as possible and it should be applied layer by layer, allowing each layer to set before the next one is applied. The plastic wood should finally protrude slightly above the level of the cabinet before it is rubbed down with fine sandpaper.

The back of the cabinet should be screwed in position using sufficient screws, but it should not be glued or it will not be easy to make any necessary adjustments.

Strengthening

It is most important that the panels be strengthened with an adequate number of supporting strips of thick wood. Many poorly constructed cabinets are almost useless owing to wall vibration at large inputs. The number of supporting struts required varies greatly with the size of the cabinet, thickness of the wood used, etc. and no definite guide can be given. If the cabinet walls are allowed to flex, then the sound output will be coloured by cabinet resonances.

Damping

The cabinet should be at least partially lined with a coating of heavy felt to subdue resonances—about $\frac{1}{2}$ in. in thickness. The felt should not be thick enough, however, to damp sound waves at frequencies below about 200c/s. It is probably wise not to use too much felt at first, as the amount can always be increased later, if desired.

No allowance should be made for the volume of the felt when calculating the cabinet volume. It is the low frequency response which is affected by the cabinet volume and low frequency sound waves can travel through the felt to the walls.

It is usual to place a piece of expanded aluminium (which is sometimes given a gold coloured finish) over the speaker and vent openings, but care should be taken that this will not produce an irritating rattle when the loudspeaker is producing sound of one particular frequency.

If a piece of cloth is used to cover the grill openings, it is important to use a type of cloth which presents little resistance to the sound waves. Closely interwoven threads should

be avoided. Cotton or plastic cloth which is very loosely woven and which has hard threads is ideal.

Finish

It is well worth while taking great care over the finish of the cabinet so that one can take a pride in its appearance as well as its acoustical characteristics.

A very neat finish can be obtained by staining the wood to the desired colour and then carefully applying a solution of shellac in methylated spirit. The shellac solution should be allowed to dry and several more coats may be applied to give the finished article a high polish.

Testing

The ultimate and most important test for any loudspeaker system is undoubtedly that of the human ear and the amount of satisfaction obtained by the owner and his friends. Nevertheless there are some measurements which can be carried out.

It is worth while to measure the loudspeaker impedance-frequency characteristic at low frequencies with the unit mounted in its cabinet. The apparatus for measuring this characteristic is given in Fig. 47, but an a.c. voltmeter should also be connected across the output terminals of the amplifier. Using low power, values of the voltage and current passing to the loudspeakers at various frequencies between about 25c/s and 200c/s can be found and the impedance at each frequency can be calculated from Ohm's law. The impedance should then be plotted against the frequency as in Fig. 45.

When the speaker is mounted in a well designed cabinet, the characteristic should no longer look like that of Fig. 45. The one large peak at the fundamental resonant frequency of the loudspeaker should have completely vanished and two

much smaller peaks on either side of the original peak should be found.

If the original peak is still present, the cabinet has not been tuned correctly. The characteristic should be replotted with blocks of wood of about 100 cu. in. in volume placed inside the cabinet. If it is impossible to remove the peak due to the loudspeaker resonance by raising the frequency in this way, the characteristic should be plotted with the vent reduced in size by a piece of wood. This will lower the frequency at which the cabinet resonates.

If peaks other than the two already mentioned are found in the response curve, of frequency between about 80c/s and 150c/s, they are probably caused by standing waves in the loudspeaker cabinet. The area of the walls covered by the damping felt should then be increased until they vanish. It is not possible to remove standing waves completely at very low frequencies, but it is possible to remove them from the speaker resonance curve.

Results

The performance of a correctly designed bass reflex cabinet system at low frequencies is much better than that of the more common and cheaper methods of mounting loudspeakers. It has many advantages, the only possible disadvantage being if the amplifier feeding the loudspeaker contains any hum, when this will sound much more prominent than before, but the remedy for this is obvious.

Care should be taken not to obstruct the vent or the front of the loudspeaker; at the speaker resonant frequency the sound is radiated almost entirely from the vent.

The frequency response should extend down to half the speaker resonant frequency or less; with conventional mounting, the bass output from a loudspeaker is almost negligible below its resonant frequency.

Design for a Radiogram Cabinet

A UDIO installations for domestic use—amplifiers, tuners, record decks, etc.—must be considered part of the furniture in most normal sized rooms, and as such require housing in cabinets which will take their place in the decor to best advantage.

For the constructor, the final design of a cabinet will be dictated by the equipment he wishes to install, which if built by himself, probably will not be of any standard dimensions. To give details of one specific cabinet then, may seem without point. However, the following design—which was built to accommodate two separate mono amplifiers, a radio tuner and a record deck—will serve as a basis from which the constructor may proceed to plan for more specific requirements.

Design Requirements

Before deciding on a design for this cabinet the following requirements were considered important.

(a) The cabinet had to be up-to-date in style to blend with the furniture already in the room.

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(b) For simplicity it had to be made with a limited number of woodworking tools.

(c) It had to compare favourably in appearance and cost with a commercially made cabinet.

The final design is considered pleasing in appearance, costs under five pounds and can be made by anyone with an elementary knowledge of woodworking tools.

Veneered Chipboard

The material used in the construction of the cabinet was a proprietary brand of faced chipboard sold in standard sizes. This chipboard is faced with African Redwood and most shops catering for do-it-yourself enthusiasts have it in stock. Would-be builders are advised to go to a shop
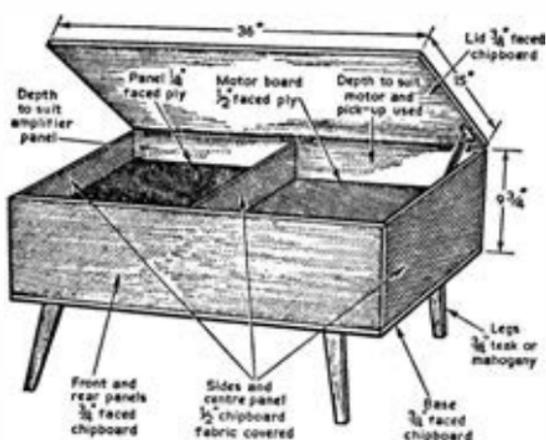

Fig. 48. The assembled cabinet prior to installation of equipment.

which has a large stock and select the four panels needed so that the veneers match as closely as possible. On purchase it is advisable to check that the lid and base panels are square by laying one on top of the other. A similar procedure is adopted for the front and rear pair and each pair should be checked for size similarity.

Whilst the size of the cabinet will have to be varied to suit different equipment, the dimensions used in this model are

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such that adequate room is available for similar units and it is unlikely that any of the standard sized panels will need to be cut. This is convenient for, if the panels were cut, the edges would have to be re-veneered. The sizes required are: 2 off, 36 in. × 15 in.; 2 off, 36 in. × 9 in.

Construction

The layout of the cabinet is shown in Fig. 48 and an "exploded" view of the main parts in Fig. 49. When a layout has been finally decided upon by the individual, suitable ventilation is provided by means of a cut-out in the base panel of the amplifier compartment. This may then be covered with expanded wire mesh.

The cut-out or cut-outs are best made by first marking the outlines on the panel in pencil, drilling a hole at each corner and cutting out with a fine keyhole saw, cleaning the edges on completion with file and sandpaper.

Next, the front and rear panels are screwed to the base leaving an $\frac{1}{8}$ in. overlap at the front and rear. Brass screws ($1\frac{1}{4}$ in. × No. 8) are used in countersunk holes and are placed approximately 2 in. apart. The resulting assembly is very strong and the screw heads are concealed.

The end panels, and also the panel dividing the amplifier and motor compartments, are cut from $\frac{1}{2}$ in. unfaced chipboard. This chipboard is cut $\frac{1}{16}$ in. smaller than the actual inside measurement between front and rear panels and is

Fig. 49. An exploded view showing the major component parts of the cabinet.

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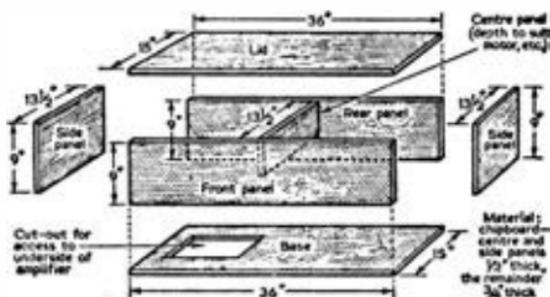
covered with fabric, preferably to match that used to face any companion speaker enclosures.

Ventilation cut-outs can also be made in these end panels, if required, the fabric effectively concealing the holes.

Compartments

The centre panel dividing the amplifier compartment from that of the record player, extends to 3 in. from the bottom of the cabinet. This allows freedom of air circulation and also for the pick-up leads and mains lead to the motor.

When completed, these panels are glued in place using one of the synthetic resin glues, care being taken to avoid glue adhering to parts of the fabric which can be seen. After glueing, the whole carcass is clamped together and the glue allowed to set. The glue should be used sparingly otherwise it will be forced out during the clamping operation and spoil the fabric panels.

Control Panels

It is next necessary to decide the layout of the amplifier and record deck panels. The original amplifiers and radio tuner were each built on chassis measuring 8 in. × 4½ in., the amplifiers each having four control knobs. Adequate depth was therefore provided by the 9 in. deep cabinet. The amplifiers and tuner were suspended by drilling the panel to correspond with the control spindles and using locknuts to hold each unit in place. Some constructors may think a piece of ¼ in. faced ply to be rather frail to accept such a weight, but this is not the case in practice.

Control Knobs

To avoid the unattractive appearance resulting from various panel designs all mounted on the one amplifier and tuner control panel, it was decided to mount the control knobs alone with no surrounding adornments. A symmetrical

layout is preferable and Bulgin K370 black pointer knobs with skirts give a professional appearance. The controls were marked with white transfers which are neat in appearance and easy to apply.

When the correct position for the control panel has been decided upon, a rail of 1 in. × 1 in. soft wood is glued and screwed around the inside edges of the amplifier compartment at the height required. This is then checked with a spirit level after which the $\frac{1}{4}$ in. mahogany-faced panel is cut to size, care being taken over this as unsightly gaps would only spoil the finished appearance.

Motor Board

A similar operation is carried out in the record player compartment, the motor board being cut from $\frac{1}{2}$ in. faced plywood. It is doubly essential to check that this board is level for obvious reasons. Sufficient clearance must be allowed above and below the board and the turntable should be positioned so that there is enough clearance all around when a 12 in. record is on the turntable. Similarly, if the motor is started by moving the pick-up outwards, sufficient room must be left for this operation. The illustrated dimensions will suit many popular record decks, but for larger transcription units, the width of the cabinet may have to be increased to 18 in.

The Lid

When the motor board has been cut to the manufacturer's template and fitted, the lid of the cabinet can be fixed by means of a piano hinge about 35 in. in length. The hinge is held by small $\frac{3}{8}$ in. brass countersunk screws. If the lid is fitted flush at the rear of the cabinet, the front edge, when closed, will be found to overlap the front panel by approximately $\frac{1}{4}$ in. This overlap does away with the need for a lifting handle.

A lid stay is fitted to ensure that no damage is caused by accidentally allowing the lid to drop.

Legs

It was decided to cut four well-proportioned legs rather than buy the wooden or ebonised legs which are available. No doubt some constructors will favour ready-made legs and these may easily be incorporated in the design.

Fig. 50. For the constructor who prefers to make the cabinet legs rather than purchase ready-made ones, details of one design are given here together with a suitable fixing block.
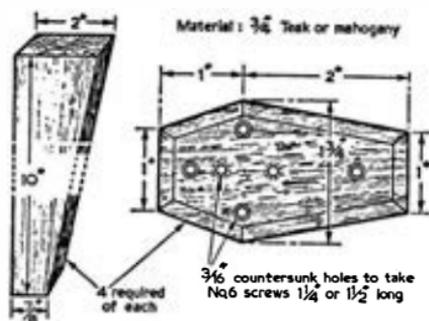
The legs are cut to the dimensions shown in Fig. 50. It is preferable to make a cardboard template and mark the four legs with the aid of this. The four fixing blocks are also cut and the legs glued and screwed to these with $1\frac{1}{2}$ in.
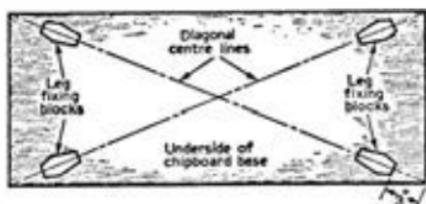

Fig. 51. Positions of the leg fixing blocks on the base of the cabinet.

countersunk woodscrews. Fig. 51 shows the fixing positions on the cabinet base.

Finishing

When the cabinet has been completed it is first rubbed down with fine sandpaper and given a coat of linseed oil. This applies to the motor board and control panel as well. When the linseed oil has dried, the whole is given a coat of

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"Lactoloid" which is rather like the clear dope used for model aeroplanes. It is recommended that this is carried out in a warm atmosphere and applied quickly and evenly with a fairly large soft brush. The cabinet is once again rubbed down with very fine sandpaper and given a final thin coat of linseed oil and allowed to dry.

A three-way connecting block is screwed to the bottom of the cabinet near the rear panel and a hole drilled alongside to take the three-core mains lead. From this block, current is taken to the amplifier, tuner and motor. A coaxial socket is fitted in a convenient place at the rear of the cabinet to connect the f.m. aerial to the tuner.


LIST OF MATERIALS

Faced chipboard panels; 36 in. × 15 in. × $\frac{3}{4}$ in.	2 off
Faced chipboard panels; 36 in. × 9 in. × $\frac{3}{4}$ in.	2 off
Plain chipboard panels; 13$\frac{1}{2}$ in. × 9 in. × $\frac{1}{2}$ in.	2 off
Plain chipboard panel; 13$\frac{1}{2}$ in. × 6 in. × $\frac{1}{2}$ in.	1 off
$\frac{1}{4}$ in. mahogany-faced plywood to suit amplifier panel	1 off
$\frac{1}{2}$ in. mahogany-faced plywood to suit motor board	1 off
Legs or wood for legs 4 off Fixing blocks	4 off
10 ft. 1 in. × 1 in. softwood	

A supply of 1$\frac{1}{4}$ in. × No. 8 and 1$\frac{1}{2}$ in. × No. 6 countersunk screws. One 35 in. piano hinge and screws. One lid stay. Fabric for end and centre panels. One 3-way terminal block. One coaxial socket. Expanded metal. Transfers, etc.

Transfers

When the amplifier and tuner have been fitted and the knobs replaced, the functions of the controls can be marked with transfers. A small pair of tweezers is useful here and the transfers should be applied after the panel has been varnished.

In conclusion it should be emphasised that the arrangement is very flexible, but whatever arrangement individual constructors care to adopt, if care is taken with the construction and finishing, the builder will have a cabinet of which he will justly feel proud.
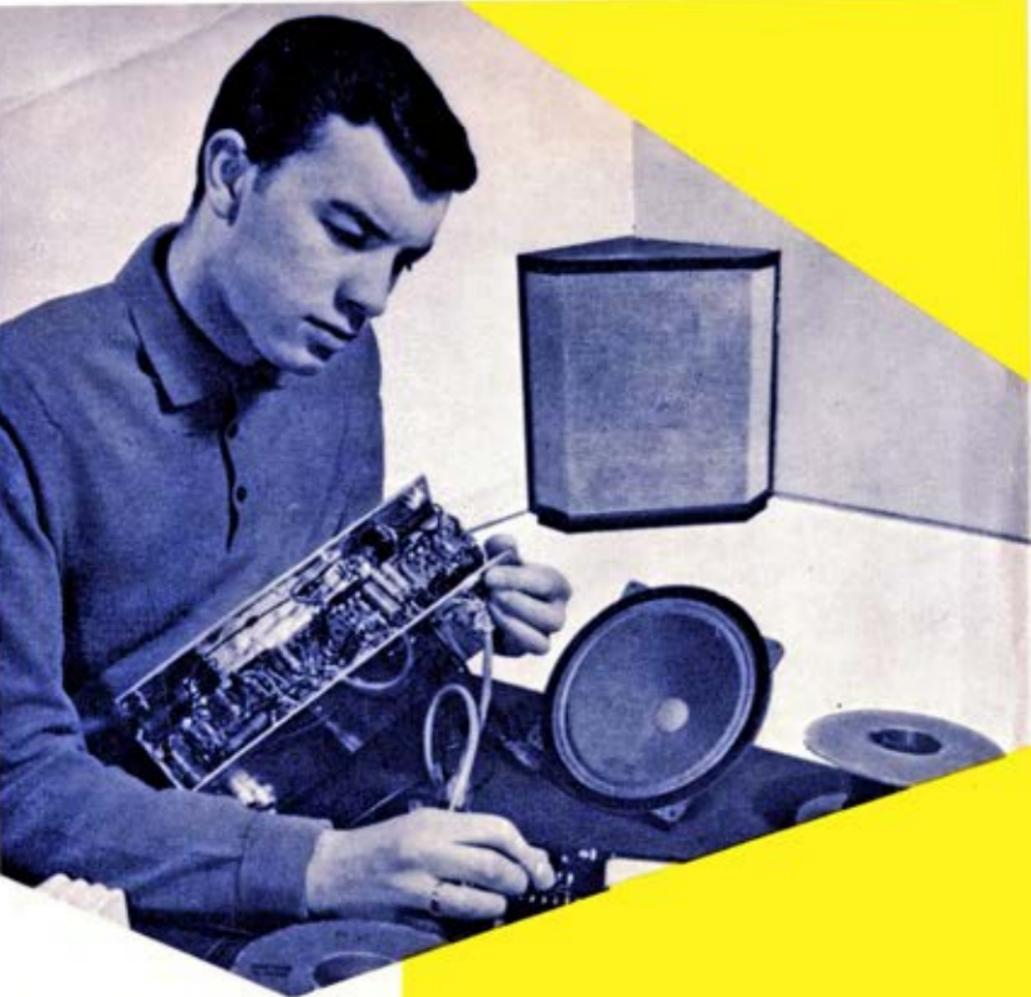
HI-FI and AUDIO...

has been specially compiled by A. T. Collins, the Managing Editor of *Practical Wireless* magazine for those with the knowledge, ability and desire to build their own equipment. An up-to-date selection of tried and tested amplifier designs for *true* hi-fi and audio enthusiasts.