A PRE-AMPLIFIER FOR

PART ONE ———————— INTRODUCTION

THE design of a pre-amplifier Control Unit presents quite a different set of problems from that of a power amplifier. From the previous series of articles, it will be seen that power amplifier design is concerned with organising a complex set of adjustable parameters. The final amplifier cannot meet the desired requirements of all parameters, and a compromise has to be reached. Thus there is always a challenge for a measurable improvement in power amplifier performance.

The design of a pre-amplifier does not contain problems of phase shift in feedback loops which include such imponderables as transformers, and almost all the design work involved can be dealt with by simple calculation, the results being easily checked. One of the great advantages in the design of a pre-amplifier as compared with a power amplifier, is that the design can be split up into stages to fulfil specified functions, and then joined up in cascade to satisfy a complete requirement.

Looking back over previous designs, it will be noticed that designs tend to fall into two patterns, the first of which is based upon the technique described above, using quite a number of stages, and in contrast, those designs which mix up complex functions around a few high gain stages—and do not make much of a job of any of them. Generally, constructors designs have been in the first category, and commercial types in the latter. This is purely a matter of economics, as it is obviously more expensive to produce a unit with more valves and circuits, than one of fewer.

However, it must be stated that, except under unusual circumstances, it has been practically impossible for the amateur to construct a control unit from a published circuit diagram, however advanced the design, to give a performance equal to the streamlined commercial models. The reasons for this are many, but are mainly due to the lack of skill in the execution, poor layout, and the inability to obtain special components which are available to the manufacturer of equipment only. Concerning application, it is difficult for the amateur to arrive at a good design from the point of layout and wiring, unless he has the time and energy to build one unit after another, at considerable expense. In addition to the difficulty in obtaining components, such as push-button switches, ganged potentiometers, polyester and polystyrene capacitors, metal oxide resistors and so on, the amateur is also faced with the problem of styling, and the finish and presentation of his completed article. He is, therefore, at a very great disadvantage when trying to compete against the manufactured item.

The kit pre-amplifier seems to be a compromise solution, but it tends to be rather bulky in order to simplify home construction for the beginner. A solution for experienced enthusiasts would appear to be the supply of a set of parts, with circuit diagram and general background information which will enable him to build professional looking equipment of advanced performance. It is the purpose of this series of articles to describe designs to a sufficient degree to enable the knowledgeable enthusiast to construct control units of such a type. The essential components will be made available later to make this practicable.

Two designs of stereophonic pre-amplifier control units will be described. One being the simplest possible no compromise design, employing four stages per channel, and another more involved design having five stages per channel, plus an additional composite channel of two stages, for three channel reproduction.

Requirements for a modern pre-amplifier

Before going into details of design, it will probably be a good thing to have a look at what we require to do.

In the evolution of any new equipment, it is normal that facilities that are not required are supplied; and also that essential requirements are not catered for. The stereophonic pre-amplifier has been no exception in this respect. The tendency in presentation today is to simplify front panels as much as possible, and to delete unwanted knobs, dials, and accoutrements. In the past, in order to be certain that every facility that the customer may require is included, some designs have been festooned with knobs and sockets, and generally—after the first flush of finding out what these things do, they are unused and serve no purpose except to clutter the panel. The designs to be described do not follow this pattern, and the guiding principle is solely one of obtaining the highest possible performance in the functions for which it is intended, as simply as possible.

The first and fundamental requirement for a pre-amplifier is to make the very best use of the signal received from high quality dynamic cartridges for disc reproduction, from an FM Tuner, and from a tape/record reproducer. It is considered that these facilities are independent, and should be capable of selection from the front panel with the units permanently wired, if desired, to appropriate sockets at the rear of the unit. The pre-amplifier control unit should be fitted with bass and treble level adjustment controls, a balance control (preferably pre-set), stereo-mono switch, a volume control, and an on/off switch.

With the advent of semi-professional three head machines with tape head amplifiers incorporating the necessary replay equalisation, there is now less need for the enthusiast to build up from a basic tape deck in order to get a good performance.

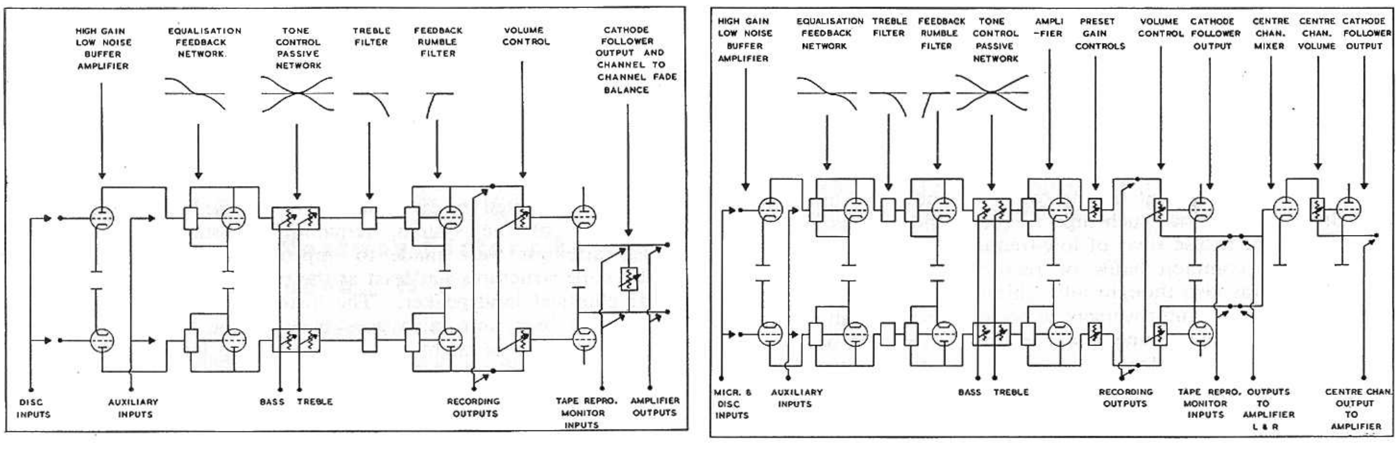
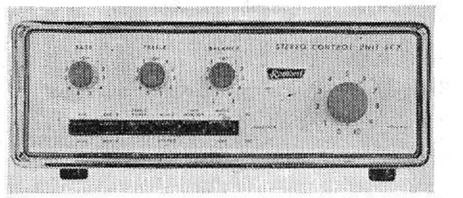


Fig. 1. *Schematic diagrams:* (*a*) *SC2 Pre-amp Control Unit;* (*b*) *SC3 Pre-amp Control Unit*

* Radford Electronics Ltd., Bristol

THE M.A. 15

by A.H. RADFORD* A.M.I.E.E.



There is now also no need for left and right channel reverse, and phase switching. Both these functions should be set correctly on initial installation and finished with. The balance control also comes in this category, but the time may not be ripe yet to dispense with it. Once the balance has been set correctly on installation, there is no need for further adjustments. Ideally, the balance should be in the form of pre-set gain controls at the rear of the unit, set on installation, and periodically checked.

Having decided on the purpose of the pre-amplifier control unit and its facilities, it can now be decided what input and output arrangements should be provided.

Reproduction from disc

At the present time most, if not all, cartridges capable of high quality reproduction from disc are dynamic, and require an input resistance of between 50 and 100K for optimum damping; the exception being the moving coil type, which has a basically low output impedance, but in practice it is matched by means of transformers, and the same input conditions apply. The transformers may be included in the cartridge itself, or fitted at the base of the arm.

Checking all the popular types of cartridges on a variety of discs with wide modulation levels, it has been found that a basic sensitivity of 5 mV is sufficient to meet all variations. This sensitivity figure refers to the r.m.s. voltage required at 1 Kc/s to fully load the power amplifiers used with the pre-amplifier. A good design figure, therefore, is of the order of 4 mV sensitivity for a standard pre-amplifier. This is the basic sensitivity of one of the pre-amplifiers described, but a basic sensitivity of 3 mV is provided in the three channel version, which is fitted with independent gain pre-sets on L and R channels.

The choice of the basic sensitivity of a pre-amplifier is an important factor. It is pointless to have a basic sensitivity of say 1 mV and compress the range of control over a short section of the volume control. Also, to reproduce signals below 5 mV, exceptional signal to noise ratio is required. It is, therefore, necessary to relate basic sensitivity to the signal-to-noise ratio. Basic sensitivities of 2/3 mV are sometimes provided, but owing to the high level of hum and noise it is unusable. This will be discussed further in the detailed design.

The requirements for equalisation for disc reproduction are now, fortunately, to one standard only, i.e., BS.1928/1961 for fine grooves, which is American RIAA.

Auxiliary 1—*Radio*. Tuners provide an output of 250/500 mV approximately. The characteristic is flat.

Auxiliary 2—*Tape Recorder*. The output from a tape machine is similarly about 200/500 mV, and an identical input arrangement is required. This also is a flat characteristic as all equalisation is logically ganged to the speed control. This is a much more satisfactory arrangement than providing compensation in the control unit.

The above input arrangements will cover 95 per cent of user requirements, and it is pointless to provide a large number of unnecessary inputs switched from the front panel; this complicates the design, and is bound either to increase the cost of the equipment or degrade the overall performance for a given amount of money. A possible additional requirement is that of a high gain facility—3 mV at flat characteristic. This has been provided in the second design, but the correction only is switched from the front panel for reasons explained in the details of design.

Tape monitor

Although not strictly an input from the amplification point of view, further input sockets are required for a tape monitoring

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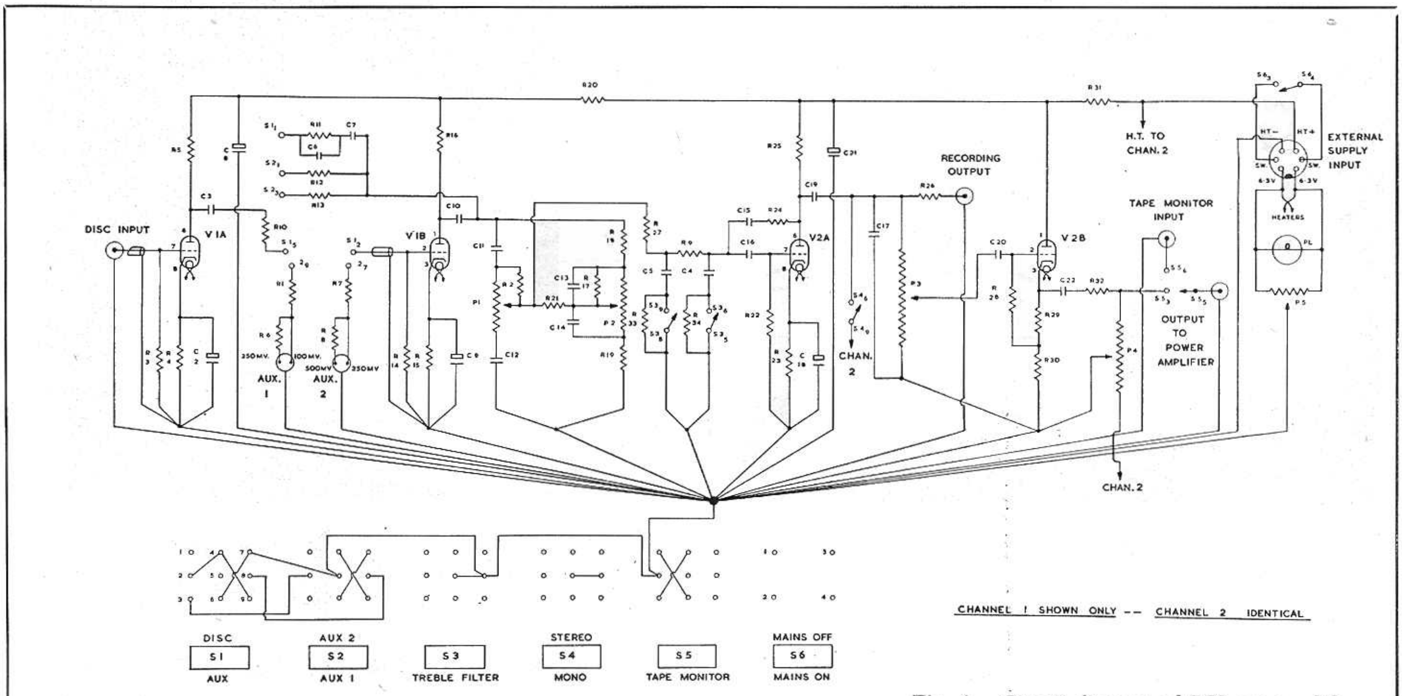

Fig. 4. *Circuit diagram of SC2 pre-amplifier*

facility. These sockets are, in fact, merely connected to a switch which disconnects the normal output from the pre-amplifier and switches the power amplifier over to the tape monitor socket. Semi-professional machines are now fitted with a third head connected to equalising pre-amplifiers which have an output of approximately 250/500 mV. With this facility, when recording from disc or radio, to tape, it is thus possible to monitor the actual recording and compare it with the incoming signal whilst recording, using the control unit for both purposes. This can be understood more fully by referring to the schematic diagram.

Output facilities

The output requirements are quite simple, being only a pair of sockets for feeding power amplifiers, and a pair to feed the input to a tape machine for recording purposes. The output level at the recording output socket is approximately 500 mV and is, of course, not controlled by the master volume control in the pre-amplifier as are the normal output sockets. It may not be necessary to mention that all input and output arrangements must be stereophonic.

Details of design

Having determined the requirements, we can now go on to discuss the detailed means of achieving this. It will be seen that a gain of approximately 100 is required in the case of disc from input to output, whereas a gain of approximately 2 is required for the high level

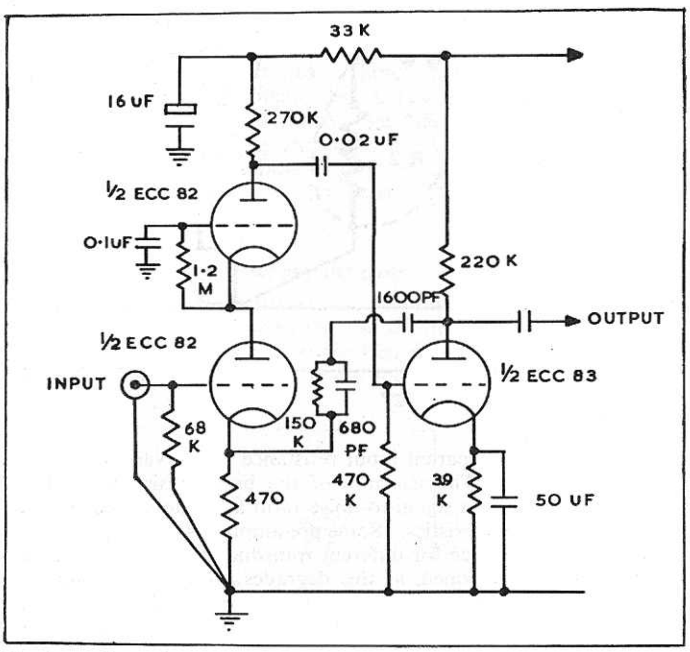

Fig. 2a. *Cascode circuit*

inputs. In order to get a gain of 100 at 1 Kc/s and include all equalisation, tone control, and rumble rejection, quite a considerable overall amplification is required. The stage requirements are basically:—amplification, equalisation, rumble rejection, and output impedance transformation. The schematic diagram showing how this is achieved in both designs is shown in **fig. 1**. The reasons for the arrangement of these designs and their analysis will now be dealt with.

Input stage

One of the most important parameters in the design of a pre-amplifier for low level disc reproduction, is signal-to-noise ratio. If a very low output cartridge is used there is no room for compromise in the design of the input stage. Nothing is more distracting than to hear hum and noise generated in the equipment as a background during passages of light modulation. The background level of professional tape machines is about −65 dB below 0 dB (0 dB is peak recording level at approximately 2 per cent total harmonic distortion). This noise is nearly all tape hiss and contains very little hum and rumble. Transfer from tape to disc does not degrade this performance very much in terms of signal to noise, if the plops and clicks are ignored. It is sensible, therefore, to aim at a performance of this order, which, it must be admitted, is not easy. It has been stated earlier that a 5 mV r.m.s. sensitivity at 1 Kc/s will give full output from the average low level cartridge from an average disc. It is therefore convenient from now on, to discuss all signal-to-noise ratio as referred to 5 mV r.m.s. from 47K ohm source resistance at 1 Kc/s. There is plenty of scope for flattery when it comes to specifying the performance of a pre-amplifier in terms of signal-to-noise ratio. It is obvious that specifying signal-to-noise ratio below maximum voltage output, as in the case of a power amplifier, is useless, but it produces a very good specification. It has been noticed also that signal to noise has been specified at 50 c/s or 100 c/s. As the RIAA characteristic has a lift of 17 dB's at 50 c/s and 13 dB's at 100 c/s, it will be seen that a vastly improved set of figures can be produced as compared with 1 Kc/s reference. Another way of making things look better is to specify the S/N ratio as at the grid of the input valve, instead of at the transducer input socket. Despite the low level of the input voltage, attenuators have to be introduced in the input circuit in some cases for reasons which will be explained in a moment, and the signal to noise ratio can look up to 15 dB's better than it actually is by this means of specification. It is the signal-to-noise ratio from the transducer itself that matters.

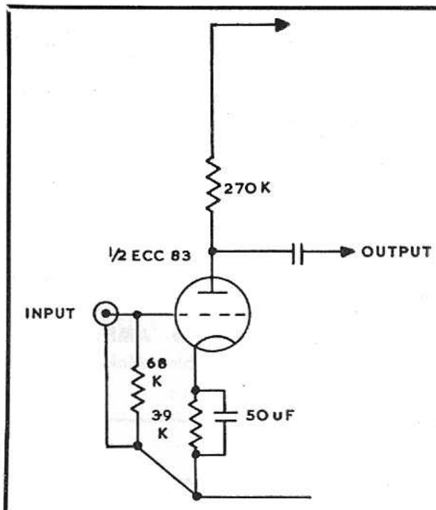

Fig. 2b. *Single triode circuit*

Practical circuits

As part of the design of the units described in this series of articles, a fundamental study was carried out into practical circuits. The cascode circuit shown in **fig. 2a** was found to have the best performance and, when using the ECC82 valve, has a signal-to-noise figure better than −80 dB below 5 mV r.m.s. with full RIAA correction with d.c. heated filaments, with a.c. heated filaments the performance is of the order of −75 dB's. The next best circuit was found to be the simple triode connection of an ECC83 (**fig. 2b**). This valve is now much better than it used to be and the modern version is free from microphony and has very low hum output. This circuit gave a figure of −77/78 under d.c. conditions and about −72/75 under a.c. conditions. The heater hum balance in both circuits is quite precise in adjustment as distinct from the EF86 valve which gives a confused hum balance. The EF86 pentode is 3 or 4 dB's below this in the triode connection, and considerably worse in the pentode connection. The ECC 83 was chosen not only because of this fact, but because of its extremely high reliability factor in service, which certainly cannot be said for the EF86. In addition to the fact that the cascode circuit contains more components, it has inherent disadvantages in other respects for this application and the straight triode is favoured.

In a practical stereophonic pre-amplifier two input stages have to be provided, and it has not been found possible to get an exact hum balance on both valves at the same time from the same heater winding. If the hum balance control is adjusted for minimum on both stages, an overall signal-to-noise ratio of −68/70 can be obtained, as against −70/72 for each channel adjusted individually.

Signal-to-noise ratios such as these are impossible to obtain when equalisation is attempted in the first stage. If it is considered that

(continued overleaf)

high signal-to-noise ratio is an important factor, then it is absolutely essential that the first stage be a buffer amplifier, with no other function but to amplify the input signal to its highest possible amount before equalisation. The reason for this is explained by referring to the circuit of an equalised input stage (**fig. 3**). Equalisation is effected by negative feedback from anode to grid, and it is important that the grid cathode impedance is not affected by the connection of the transducer, as the feedback would then be modified, affecting the equalisation characteristic. To overcome this a series resistance is necessary to buffer off the impedance of the transducer, and it will be seen from this that an L type attenuator is formed by the series resistance and the effective grid input resistance. The effective input resistance R.2 is made up of the actual grid resistor

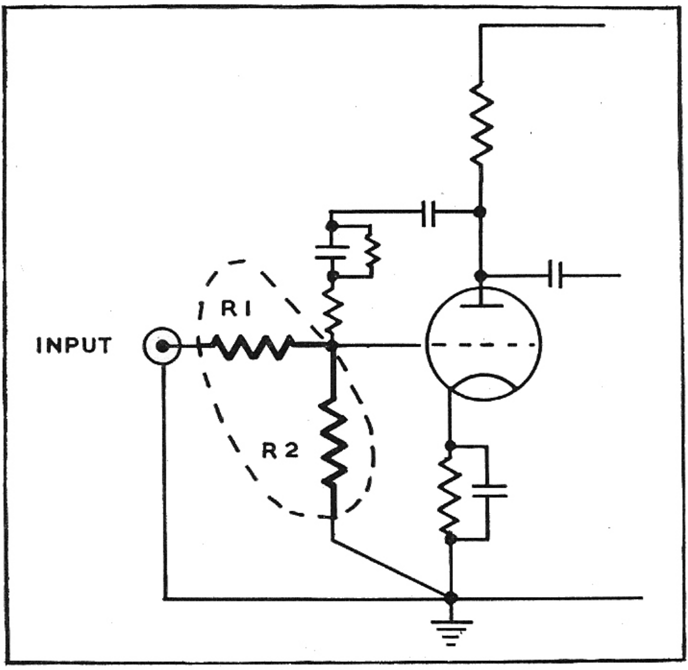

Fig. 3. *Equalised input stage*

and the very low apparent input resistance of the valve due to feedback, in parallel. The selection of the buffer resistance R.1 is a compromise between signal-to-noise ratio and interference with the equalisation characteristics. Some pre-amplifiers have the facility of varying this resistance for different transducers for optimum results. As previously mentioned, as this degrades the transducer signal-to-noise ratio, it is advantageous in this case to specify a hypothetical signal-to-noise ratio at the grid of the valve instead of at the transducer input, if a satisfactory looking figure is required.

Valve circuit most suitable

It is rather an interesting reflection that the simplest and oldest of valve circuits is still the best for this application. It is of interesf also to note that it is used as the first stage head amplifier in one ot the best known professional tape machines. The anode resistor must, of course, be a metal-oxide low noise type to obtain the figures quoted.

At the present time the lowest noise transistor is quite out of the running for this application.

From an application point of view, it is essential that the input socket be connected directly to the grid of the buffer amplifier through a screened lead with no switching. Switching at this level inevitably picks up hum, noise and crosstalk.

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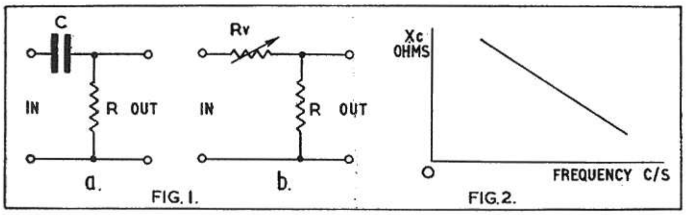

Fig. 1. *Simple R.C. network.* **Fig. 2.** *Reactance frequency relationship for capacitance*

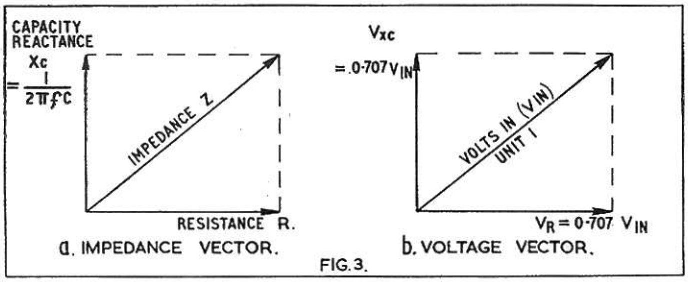

Fig. 3. *Impedance and voltage Vector diagrams*

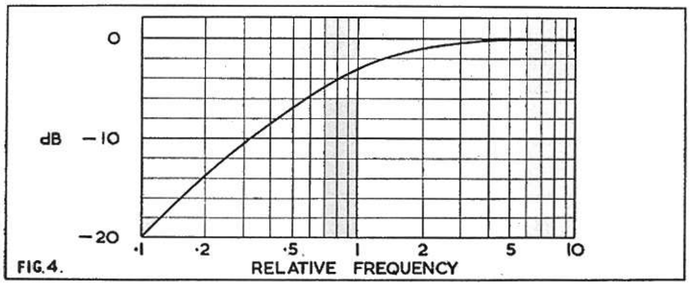

Fig. 4. *Input output relationship versus frequency for network of* **Fig. 1a**

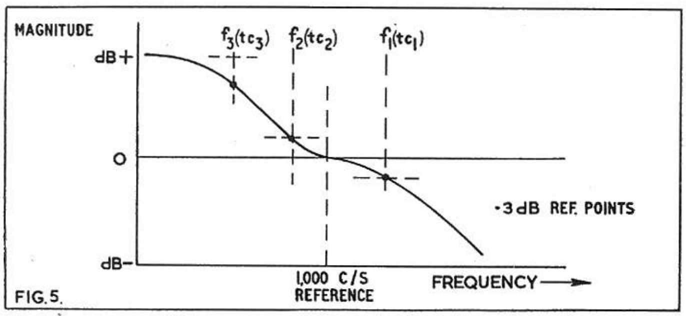

Fig. 5. *Basic equalisation curve*

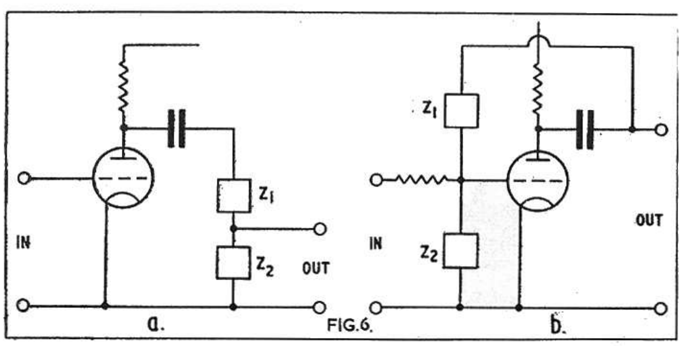

Fig. 6. *Basic equalisation circuits: (a) passive; (b) active*

THE PRE-AMP

BY A. H. RADFORD, A.M.I.E.E.

FOR reasons which are well known, it is advantageous to record to a frequency/amplitude characteristic which is not a straight line over the frequency band desired. Since the beginning of recording, many characteristics have been used and these have differed between countries, and between individual companies, making equalisation within reasonable limits of accuracy impossible.

The RIAA characteristic which may now be considered as standard, is specified in the British Standards Institution booklet B.S.1928/1961. Two characteristics are described; for coarse grooves (78 r.p.m.), and fine grooves (45 and 33⅓ r.p.m.). The coarse groove is no longer of any interest in the design of new equipment because of its obsolescence, and will not be considered further. Adjustments to the variable tone control system can be made if desired to correct for small differences between recording characteristics.

Before detailing the desired reproducing characteristic it will be as well to examine the fundamental basis of equalisation and to see how different frequency/amplitude characteristics are obtained.

Fig. 1 shows a simple R.C. combination for coupling one stage of a valve amplifier to another. It is also a frequency sensitive network. For the purpose of illustration, for A.C. signals the capacitor " C " may be replaced by a variable resistor R_v, whose resistance is a function of the frequency of the signal being transmitted from " *in* " to " *out* ". It is obvious that the higher the value of the resistance R_v the smaller will be the signal appearing at " *out* " from " *in* ".

This imagined resistance of the capacitor " C " is called *reactance* (X_c) and is $\frac{1}{2\pi fC}$ ohms, where the frequency(f) is in c/s and the capacity (C) is *Farads*. It will be seen that X_c is inversely proportional to frequency, and a reactance curve as in **fig. 2** is produced. Thus less and less of the signal is transmitted through the network as the frequency is lowered, and therefore it could qualify as a " bass cut " circuit. In normal R.C. coupling circuits the capacitor is made large enough such that the reactance X_c is very small in comparison with the termination resistor R at the lowest frequency it is desired to transmit, and a negligible loss occurs in the desired frequency band.

When C equals R

The frequency at which the reactance X_c of the capacitor " C " equals the resistance of the load R is the important issue at this juncture, and is known as the " 3 dB reference " or " ½ power " point. It is necessary at this stage to qualify this, for unless a little knowledge of the A.C. theory is in the background, it can appear confusing.

Take now the circuit of **fig. 1b**. If $R_v = R$, then the voltage appearing at " *out* " is half the input voltage, and the attenuation is therefore $20 \log_{10} \frac{V\,in}{V\,out} = 20 \times 0{\cdot}3 = 6$ dB—not 3 dB! The reason for this is that in the case of the capacity reactance " X_c " the capacity voltage is out of phase with the voltage appearing across R and cannot be added or subtracted arithmetically. The voltages across pure capacitance and resistance are always 90° out of phase, and the combination voltage phase some way in between, depending on the relative magnitudes of reactance and resistance. Such quantities can only be manipulated vectorially, and the impedance is given by the vector sum of reactance and resistance. See **fig. 3.** **(3a)** shows an impedance magnitude vector and **(3b)** shows its equivalent voltage vector.

The relative values can be obtained graphically or by simple trigonometry. In the case where $X_c = R_1$ and the input voltage V_{IN} is considered as unity, it will be seen that the voltage across

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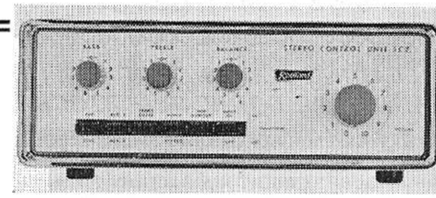


PART TWO — EQUALISATION

R is 0·707 V, and not 0·5 V as in the simple resistance network. The attenuation thus becomes 20 $\log_{10}$ 1·414 = 20 × 0·15 = 3 dB.

It is well to observe that, if the voltage is reduced to 0·707 of its former value, the output current, assuming the load resistance remains unchanged, will also be reduced proportionally. The power transmitted is $V \times I$, and is now $0{\cdot}707^2 = 0{\cdot}5$. Hence the 3 dB reference = ½ power point.

It is now apparent that the output voltage/frequency relationship can easily be worked out for any value of resistance and capacitor network, and the attenuation curve plotted as shown in **fig. 4**. So much for the fundamentals. We can now proceed with the design of an equalisation stage.

Referring to **fig. 5**, which is the basis of B.S.1928, it will be seen that not one 3 dB point, but three, are included. This looks complicated, but is actually three simple time constants which can be calculated independently.

All equalisation circuits are basically loss circuits, in which the losses vary in accordance with the desired characteristic. If **fig. 5** is looked at from the 1 Kc/s reference, then an apparent gain is required as the frequency is lowered. The amount of this gain

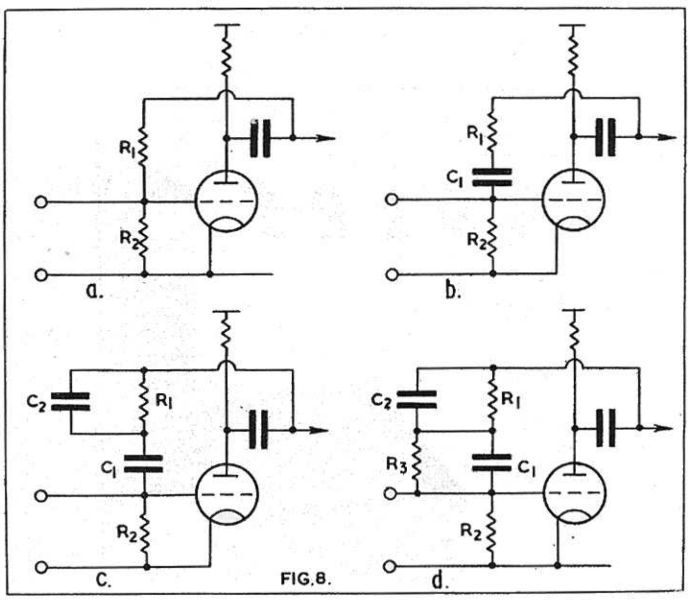


Fig. 8. *Detailed equalisation circuits: (a) Feedback only (flat characteristic); (b) Equalisation for f_2 3 dB reference; (c) Equalisation for $f_1 + f_2$ 3 dB reference; (d) Equalisation for $f_1 + f_2 + f_3$ 3 dB reference*

must, of course, be inserted as a loss at 1 Kc/s and recovered at the low frequency.

Equalisation may be effected as a passive network as in **fig. 6a**, or in an active network as in **fig. 6b**. In the passive case the output of the valve is lost into the network, and in the active case the frequency sensitive network controls the gain of the stage, to produce the characteristics required. Fixed recording characteristic equalisers are most frequently of the active type, but passive networks are used as variable equalisers as well as active types. The full RIAA (B.S.1928) characteristic for reproducing apparatus is shown in **fig. 7**.

Proceeding with the detail design, using feedback equalisation, we start with the circuit of **fig. 8a** and decide: (*a*) The approximate values of R_1 and R_2 for the 1 Kc/s reference feedback amount. These resistance values are best determined empirically, and they need to fit standard values of capacitance for equalisation. The values will, of course, be controlled by the gain of the valve used. Taking practical cases, suitable values of R_2 for optimum feedback have been found to be 680 K ohms for a triode and 100 K ohms for a pentode. This leaves only R_1 to decide upon for feedback amount. The R.C. product is then simple as detailed below.

(*b*) Referring now to **fig. 7** the 3 dB bass lift reference point (f_2 on **fig. 5**) is at 500 c/s and the reactance of C_1 should be equal to R at this frequency, i.e., $X_{c1} = R_1$ at 500 c/s (**fig. 8b**).

(*c*) Similarly at f_1 2,120 c/s is the top cut 3 dB reference point. This is obtained by increasing feedback with frequency by shunting R_1 by a capacitor C_2. Thus $X_{c2} = R_1$ at 2,120 c/s (**fig. 8c**).

(*d*) If a large amount of feedback is used, then it will be necessary to limit the increase in reactance of C_1 by a resistance across it (**fig. 8d**). In this case $X_{c1} = R_3$ at (f_3 = 50 c/s). However, if the feedback at 1 Kc/s is made the same as the lift required at 30 c/s, the step resistor R_3 is unnecessary, since the stage then gives maximum gain at this frequency, and the response flattens.

Another way of arriving at values of C_1, C_2 and R_3 is from the specified time constants. Specifying the time constant is just another way of giving the R.C. product and frequency. B.S.1928 gives the time constants for f_1 as 75 μsecs., f_2 as 318 μsecs., and f_3 as 3,180 μ secs.

For example

t_2 = 318 micro-secs. = C_1R_2(f_2 = 500 c/s)
t_3 = 3,180 micro-secs. = C_1R_3(f_3 = 50 c/s)
t_1 = 75 micro-secs. = C_2R_1(f_1 = 2,120 c/s)
(where C = pf, R = M ohms)

As suggested, the design is largely empirical and the final values are found by test in the actual circuit arrangement used. It is essential that the characteristics achieved are not unintentionally modified by the resistance of the source and load. A series resistance is used to isolate the equalisation stage from the signal source resistance. This was discussed in the previous article in respect of signal to noise ratios. By careful design it can be arranged, if desired, that the C.R. values of the subsequent stage provide the shelving of the low frequency response at 50 c/s. This is done in the two pre-amplifiers described. The actual circuit values used for the time constants are the readily available " preferred " values

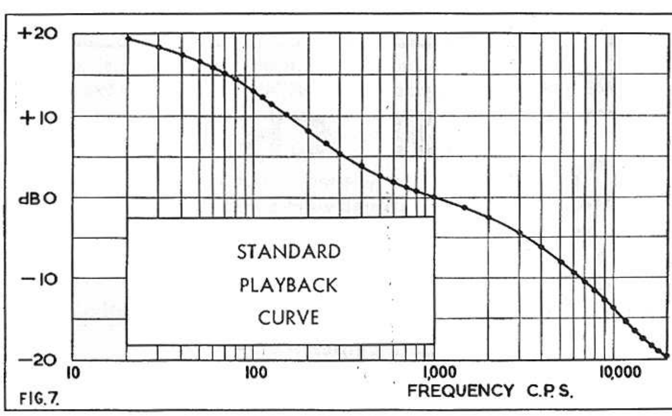


Fig. 7. *RIAA (BS 1928) Characteristic with plotting points*

(*continued overleaf*)

of 5% tolerance as follows:—

SC 2 Pre-amplifier Control Unit

t_1—$R_{11}.C_6$ = 390 K and 200 pf = 78 micro-secs. (BS. 75 micro-secs.)

t_2—$R_{11}.C_7$ = 390 K and 910 pf = 354 micro-secs. (BS. 318 micro-secs.)

SC 3 Pre-amplifier Control Unit

t_1—$R_8.C_7$ = 270 K and 270 pf = 73 micro-secs. (BS. 75 micro-secs.)

t_2—$R_8.C_6$ = 270 K and 1,200 pf = 324 micro-secs. (BS. 318 micro-secs.)

The response curves obtained using these values in production are within 2 dB of the specified curve.

The principles outlined above apply to any equalisation or tone control system, and if thoroughly understood any practical system can be worked out with the minimum of effort. For those who like a little brain fag, the curve is specified as follows:—

$$N(dB) = 10 \log\left[1 + \frac{1}{4\pi^2 f^2 t^2_2}\right] - 10 \log(1 + 4\pi^2 f^2 t_1^2) - 10 \log\left[1 + \frac{1}{4\pi^2 f^2 t_3^2}\right]$$

where f = frequency in c/s and the time constants t_1, t_2, t_3 being as specified below for fine grooves:

t_1 = 75 micro-secs.
t_2 = 318 micro-secs.
t_3 = 3,180 micro-secs.

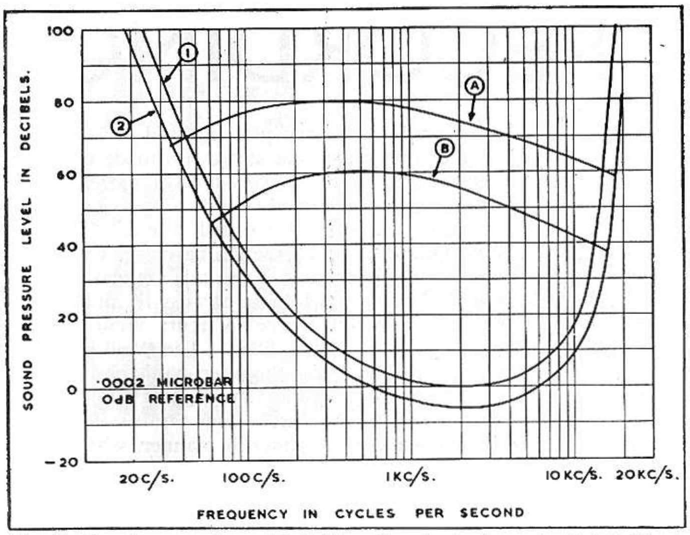
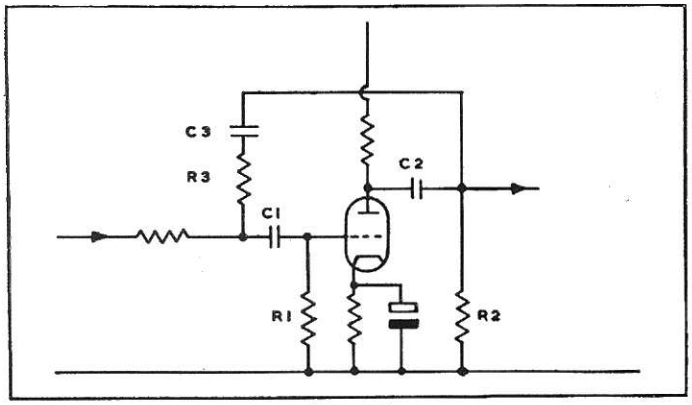


Fig. 1. *Loudness contours (1 & 2) and orchestral spectra (A & B) at two levels*



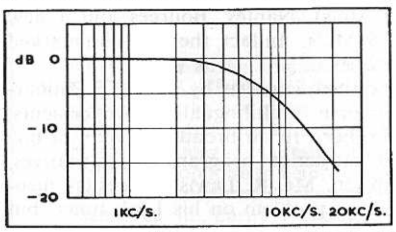


Fig. 2. *(above) shows the circuit used to provide a sharp fall-off at low frequencies, the various components contributing as in fig. 3 (below)*
Fig. 5. *on the left depicts the response of treble filter in fig. 4*

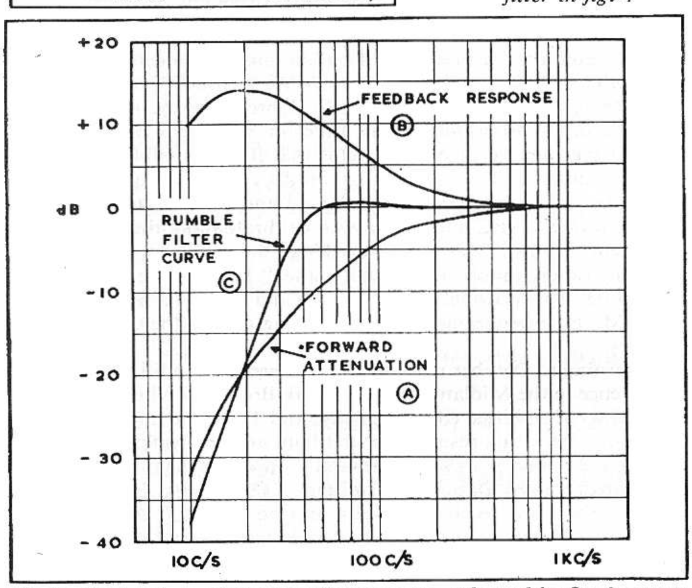


Fig. 3. *Derivation of high pass filter response achieved by fig. 2 circuit*

THE PRE-AMP

BY A. H. RADFORD, A.M.I.E.E.

IN the last two instalments we have dealt with the requirements of signal-to-noise ratio and equalisation. The subject matter of this article is concerned with filters and tone control.

Before discussing detail design, it will probably assist if the requirements are outlined. They concern:

(*a*) The attenuation of very low frequencies (LF cut off, or rumble filter)
(*b*) The attenuation of high and very high frequencies (treble filter).
(*c*) The frequency characteristic of the system in respect of the overall balance of low to high frequencies about a mid-frequency point (tone control).

Almost a Necessity

It is now generally agreed that a rumble filter is almost a necessity for the high quality reproduction of sound. If the filter is a good one and does not degrade the audible bass response, it should be a permanent feature, but if it is a simple type with a poor cut-off response (as in low and medium priced equipment) it should be switchable, in or out. Popular literature suggests a low frequency audible limit of 20 c/s, and this is a figure which has been used for the design of power amplifiers for many years. This, however, is a simple generalisation. It is necessary to examine the subject more thoroughly in order that the filter cut-off frequency may be set as high as possible with the maximum possible rejection of unwanted frequencies.

Unwanted frequencies may be defined as:

(*a*) Frequencies which were not in the original performance, and due mainly to mechanical imperfections in recording and reproducing equipment.
(*b*) Frequencies which were in the original performance, but so low that if it were possible to reproduce them, they would still be inaudible, and if an attempt were made to reproduce them, audible distortion would result.

Suitable Cut-Off

It would appear then that a suitable cut-off frequency would be the lowest frequency that can be reproduced satisfactorily, and yet be audible. From experiments and experience over many years, the writer has formed the impression that the limit of hearing under normal listening is around 40 c/s, and no advantage could be obtained by the inclusion of lower frequencies, even if loudspeakers could be made to reproduce them satisfactorily. Recently, in order to obtain more information about these boundary frequencies, a transducer was constructed to radiate a pure acoustic sine wave at a level equivalent to 10 watts into a reproducer of 2% efficiency (0·2 acoustic watts). It was found that at 40 c/s the sound level to the ear appeared loud and satisfying, but the intensity judged by ear at the same acoustic pressure dropped catastrophically with frequency, and 30 c/s was inaudible.

Allowing for the Ear

It is well known that the sensitivity of the ear varies considerably with frequency, particularly at the low and high ends of the spectrum. **Fig. 1** illustrates this, and is a diagram showing how the threshold of hearing response varies with frequency. Curve (1) is for an average listener, and curve (2) for an audio engineer or experienced hi-fi enthusiast. If the curve for critical hearing is referred to, it will be seen that an acoustic pressure increase of about 14 dB is required to make a 30 c/s tone audible as compared with a 40 c/s tone. This is a power ratio of 25. Assuming a power of 4 watts is required for the satisfactory reproduction of a 40 c/s tone, then

FOR THE M.A.15

PART THREE - FILTERS AND TONE CONTROL

an amplifier of 100 watts output is required for equal audibility at 30 c/s.

Superimposed on the threshold level curves is a curve (*A*) showing the approximate distribution of intensity from a symphony orchestra at 100 dB peak level above 0·0002 microbars (about 20 ft from the orchestra). It will be seen that even at this intensity of sound, the level at 40 c/s is barely above the threshold of audibility. Curve (*B*) is a reasonable reproduction listening level, and indicates that only frequencies above 60 c/s have any significance. From this one can see that the apparent audible bandwidth is a function of the listening level, and that reproduction down to 40 c/s is adequate even at symphony orchestra levels.

Curves are published by loudspeaker unit manufacturers showing responses down to 20 c/s. This appears to be a matter of convention; these curves are arrived at in a totally unrealistic manner, and do not in any way indicate the performance at listening levels. The curves are taken either (*a*) at a microscopic input, with the microphone very close, such that the diaphragm assembly is not limited by non-linear movement at low frequencies, or (*b*) operated at normal listening level, such that the distortion at low frequencies

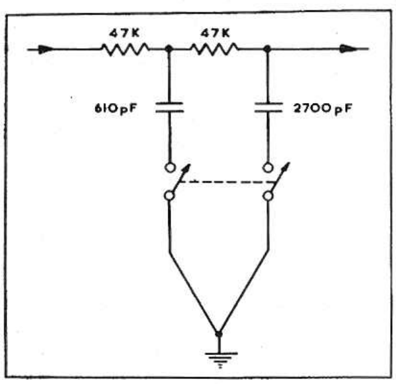


Fig. 4. *Low pass filter circuit with controlled rate of slope as depicted in fig. 5. This is preferred to a circuit with very sharp response for reasons given in the article*

is equal to what the fundamental might be, if it were present; ie., no recognition is paid to the fact that if a 30 c/s signal is being fed in, then the acoustic output being measured is frequently comprised almost entirely of harmonics: 60, 90, 120 c/s etc., and other unrelated frequencies.

If frequencies which are not part of the original music programme are fed to a loudspeaker, then not only will the speaker produce harmonic distortion, but intermodulation distortion, and other spurious frequencies. If the loudspeaker is of the bass reflex type, transient distortion will also be introduced, depending on the acoustic damping, as the mass of the speaker system and the enclosure " rings " when excited by distortion products and intermodulation from inaudible frequencies. This is exhibited by a general low-frequency background during quiet passages, and a confusion of low frequency sound at a reasonable listening level.

In the two pre-amplifiers described, full amplification is maintained down to 40 c/s, and then the gain is attenuated as rapidly as possible. This gives a turnover frequency of 35 c/s (3 dB reference) and an attenuation of between 30 and 40 dB at 10 c/s. It would be of considerable advantage if recording and broadcast companies used a similar filter at every opportunity. Even in the transfer of already filtered programme material from tape to disc, another filter is desirable to eliminate some of the unwanted frequencies which the pick-up and pre-amplifier have to cope with. It cannot be overstressed that the frequencies which it is desired to eliminate, i.e. below 40 c/s, are under practical circumstances inaudible. It is the distortion in amplifiers and loudspeakers which generates audible and objectionable spurious frequencies when fed with these sub-audible frequencies.

Listening tests carried out with filters of different cut-off frequencies over a considerable period of time, on loudspeakers with exceptional low frequency response, on all kinds of programme material, at various sound intensity levels, have produced these conclusions:

(1) A low frequency cut-off of 70 c/s does not audibly affect the sound quality at average listening levels on a large proportion of programme material.

(2) A low frequency cut-off of 35 c/s does not audibly affect any programme material at any level.

(3) In all cases a low frequency filter improves listening by cleaning the desired low frequency response and reducing the general background.

A Suitable Circuit

The circuit of a suitable rumble filter is shown in **fig. 2.** This is a straightforward negative feedback amplifier, with the time constants adjusted to achieve the response as shown by curve (*C*) of **fig. 3.** The operation of the filter is briefly as follows: C1, R1, and C2, R2, form a high pass filter which has an ultimate slope of 12 dB per octave, as indicated on curve (*A*) of **fig. 3.** Negative feedback bass lift is applied by means of C3 and R3, which produces a response similar to that of curve (*B*) **fig. 3.** The combination of curves (*A*) and (*B*) produces the desired curve (*C*). These are simple time constants, and can be calculated in the same manner as the constants required for equalisation described in the last instalment, or more simply read from the abac printed in this issue **(fig. 8)**.

There is good scope for experimenters in designing and testing filters of this kind, and many interesting hours can be consumed in so doing.

The performance characteristics of *treble* filters incorporated in most pre-amplifiers are, unfortunately, a relic of history rather than a reflection of today's requirements. It has as its background the desire to remove unwanted noises, such as heterodyne whistles and " needle scratch ", to use old terminology. In order to do this, it was thought that if a very sharp filter were used, then the offending frequency, or band of frequencies, could be removed without affecting the reproduction of the desired frequencies in the pass band. It was also thought that surface noise was mainly a high frequency function, and that improved reproduction could be obtained by

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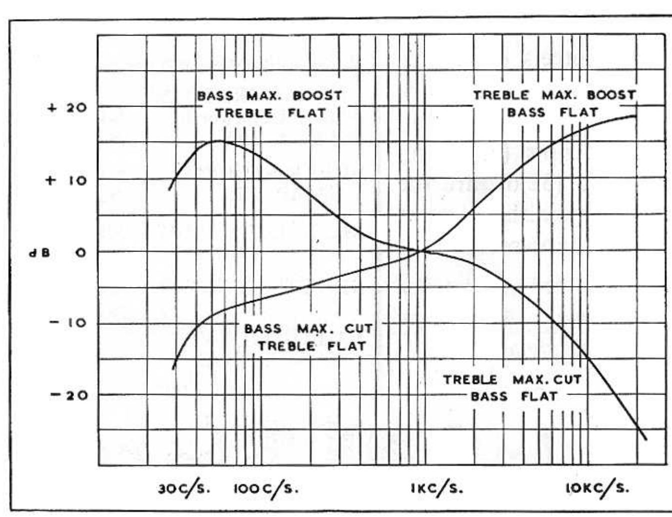

Fig. 6. *Limits of bass and treble control range in pre-amp circuit*

accepting all frequencies up to a specified figure, and then cutting the response off sharply. This we now know to be false, as surface noise has random distribution, and differential selection against non-musical frequencies cannot be made by sharp cut filters. Instead, as will be explained later, things can be made worse.

A requirement of more recent date, for a high frequency filter, is for the alleviation of distortion products. If non-linearity exists in any transmission system, distortion products will be introduced. For some reason, it is assumed that all distortion products are at high frequencies, presumably as some of these frequencies are multiples of a fundamental note in the mid range. Here again, the sharp cut filter fallacy was perpetuated, and as would-be purchasers had grown accustomed to believing this, manufacturers fitted them willy-nilly as a sales necessity. In order to placate the more enlightened, and to make it appear that a filter would satisfy all requirements, the filters have been made variable, not only in respect of the cut-off frequency, but also the attenuation rate.

In a non-linear system, distortion products of simple harmonic relationship occur over the whole of the frequency band, which are not necessarily objectionable. Not only do we have simple harmonic frequencies of the transmitted fundamental frequency added, however, but intermodulation products also. With the " beating " of all these frequencies, a sound which would normally give pleasure soon becomes objectionable. As with surface noise, distortion products have a wide range, and no advantage can be gained by passing all frequencies up to a certain point in the scale, then cutting off all above it.

Owing to a function known as transient distortion, a sharp cut filter, depending on its slope, will frequently produce a more objectionable sound when in circuit, than out. Although a large slice of distortion products in the programme material is eliminated, the transient distortion introduced by the filter can outweigh its advantage. Under these circumstances, not only is a vast amount of programme content lost, but distortion products and background noise can be increased. All networks have this characteristic to a degree, and it is a function of the slope, or the " Q " of the circuit. It does not matter how the slope is produced, whether by lumped reactances or simple RC in a feedback circuit; the result is the same. The frequency/phase response below the cut-off frequency is drastically modified at different frequencies and the start and decay characteristic of a complex music waveform is changed. The ear is particularly sensitive to wave-front disturbance.

This distortion is also referred to as transient ringing, transient coloration, or phase distortion. When a sharp cut filter is excited by a steep wave front, whether it be due to programme material or a distortion product, it can be made to " ring " exactly as a tuned circuit. Thus, whatever signal is fed into the filter, it will feed energy back into the circuit at its resonant frequency and modify

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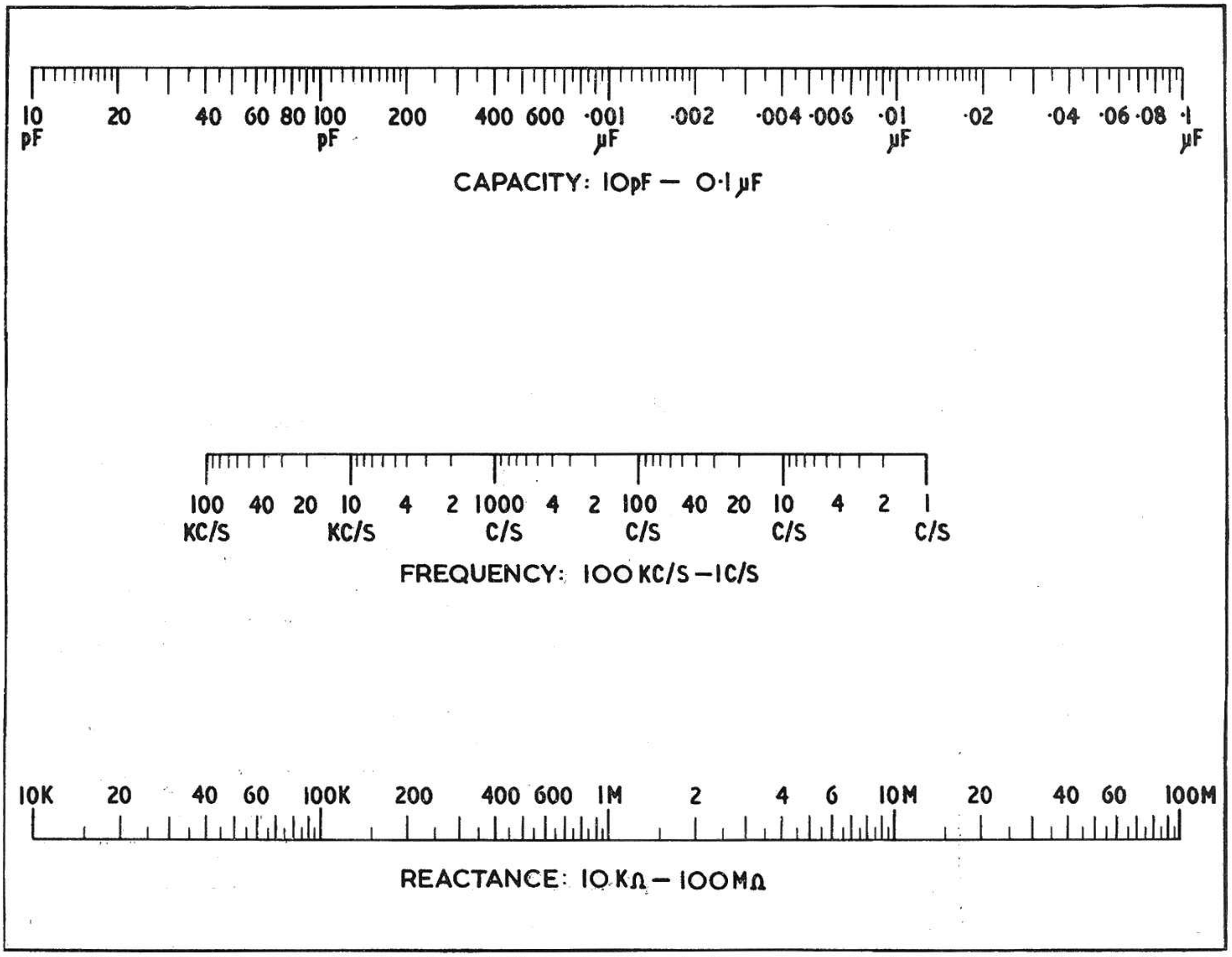

Fig. 8. *This abac facilitates quick calculation of capacitative reactance at audio frequencies, or gives the capacitor value for a given frequency and reactance. Simply place a rule across the chart between two known quantities and read the unknown value on the third line; e.g., a rule between 0.02mfd and 100c/s gives 80K ohms. For capacitor values above 0.1mfd, take a figure of, say, one-tenth of the wanted value and divide the resulting reactance by ten; e.g., 0.04mfd equals 20K ohms at 200c/s, which means that 0.4mfd will equal 2K ohms.*

the phase characteristic. This can be so bad as to give the effect of a " hollow tube " coloration to the programme material. In addition, the spurious frequencies produced will cause further distortion products with existing programme material, distortion and surface noise, and so give an apparent increase in surface noise and distortion.

It is true that distortion appears more objectionable in the high frequency region, but it should be realised that if distortion products and noise are present in programme material, they cannot be eliminated; but an alleviation can be effected by reducing the general overall high frequency response. It is simple to test a filter with pulses, step functions, and tone bursts, but as the ear and the brain finally decide what is acceptable and what isn't, this information can only be of use after subjective tests with programme material and " white noise ". Experimenting with the shape of these filters in subjective listening, it would appear that a considerable slope can be reached ultimately, provided it is a gradual roll-off with increasing slope. In other words, the slope rate can be increased with frequency within certain limits, without transient distortion. It is not possible to discuss this in detail in this article, but it is sufficient to illustrate an attenuation curve which has been found to provide the maximum alleviation of distortion products, with the minimum of interference with the programme material, in a wide range reproducing system. This attenuation slope used in conjunction with a suitable tone control system will enable a listener to deal with any usable pro-

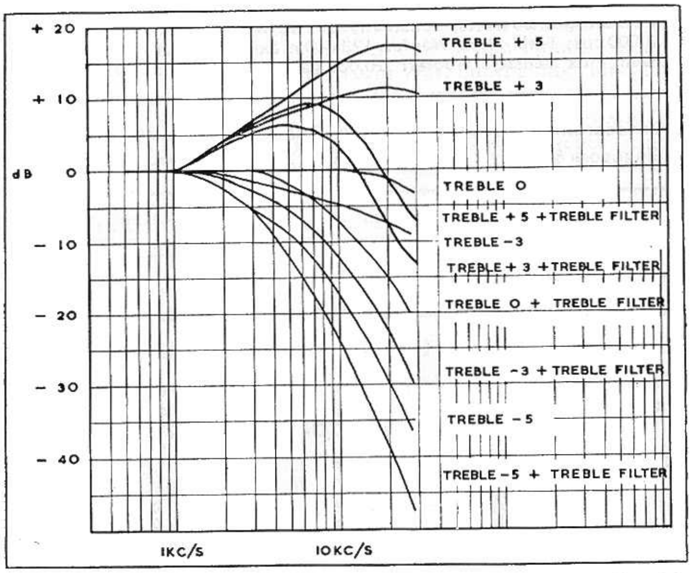

Fig. 7. *Various treble responses obtainable using filter and treble control*

gramme material. This filter slope is obtained by a two section RC network, shown in **fig. 4**, and its response in the pre-amplifier in **fig. 5.**

Turning now to the tone control, here again, it is desirable to re-state the requirements of a tone control system and its function in a pre-amplifier control unit. The basic requirement is to adjust the general level of low and high frequency response about a mid-frequency. The amount of lift at low and high frequencies which can be used under normal conditions is very limited. It is difficult to correct for transducer weaknesses by the tone control, but a general tendency may be corrected, i.e. if a reproducer has a closed acoustic system and a highly damped bass resonance, it will be found that raising the general level of low frequencies will improve the balance. The tone control should be of use, also, to obtain a satisfactory balance between low, high, and middle frequencies, depending on the acoustic level of the reproduction. Variations of balance exist in programme material, particularly on discs, which may also be improved to some degree. It should be emphasised, however, that no startling improvements can be expected by adjustment of the tone control system in this respect, but slight adjustment is frequently worth-while, particularly in the case where over-emphasis has been given to high frequencies in the recording process.

Generally speaking, a lift of 8 dB at 50 c/s and 10 Kc/s is the maximum likely to be required, and with a wide range reproducer lift is rarely desired at high frequencies, treble attenuation being more frequently required. The tone control system in one of the pre-amplifiers described has a variation of ± 10 dB, except for bass cut which is a rare requirement. The SC 3 has a variation of ± 15 dB, as shown in **fig. 6.** The tone control circuit in both SC 2 and SC 3 is the same, although component values are different. The circuit has been in use for some considerable time, and was specified in the original Mullard pre-amplifier circuit, and is frequently used in commercial pre-amplifiers. The circuit constants may be modified to suit the desired characteristics from the principles outlined in the last instalment. Some difficulty can be experienced, however, with incorrect values, in obtaining a flat response when the control is set to its mid position.

This circuit is preferred to a negative feedback tone control system which was popular some time ago, the reason being that the whole level of the bass and treble response about a constant mid frequency point is variable, whereas feedback systems tend to produce a variable crossover point, which is undesirable.

In cases of " over brilliant " recording with a large number of distortion products, the tone control may be used in conjunction with the treble filter. A considerable variation in the reproduced characteristics can be obtained by using the two controls, which will cover all usable requirements. A graph showing the variations which can be obtained is given in **fig. 7.** It will be seen also that if the treble filter is utilised with the tone control in treble lift position, a so called " presence " effect is obtained.

It is well to remember that the use of the tone control, other than in small amounts, is a pathological state, and if it is found that wide variations of the tone control are required to produce a pleasing sound, then a fundamental weakness exists somewhere. Using wide range transducers, it will be found that on the *best* recordings of today, the most satisfying results will be obtained with the controls near the flat position, and without treble filtering.

Next month the final instalment will discuss the output stage, three channel reproduction, and give component values and other details.

THE PRE-AMP FOR THE M.A.15

BY A. H. RADFORD, A.M.I.E.E.

Some Final Constructional Notes

THE first requirement in a stereo pre-amplifier is that unwanted feedback is not present in either channel, or between channels. The presence of unwanted feedback in an individual channel may produce an unstable condition; the amplifier may be on the verge of instability or, in fact, in oscillation at a supersonic frequency. Stability is one of the most important parameters in an amplifier system, and it is the main factor in the situation where two amplifier systems of apparently similar performance specifications sound quite different.

The design of the pre-amplifiers described is inherently stable from the circuit point of view. All circuits, except the input valve stage, are operated at low impedance, due to negative feedback, and capacitance effects between stages are therefore reduced to small proportions. As previously explained, most of the gain in the pre-amplifier takes place in the input stage, and an extra amount of care must therefore be taken here. The input stage is used for discs, and in the case of the SC3 for microphone also. All high level inputs are taken to the second stage, where the tendency for instability, owing to the reduced overall gain, is not so great.

Unstable Tendencies

If a pre-amplifier has a tendency to instability, it is not always easy to detect unless a considerable amount of equipment is available. If ringing is apparent on a 5 Kc/s square wave, then some effort should be made to improve the layout. Treble lift will tend to increase HF instability. If equipment is not available, instability may be detected by a sudden increase or decrease in the noise level when increasing the treble lift; sometimes a " plop " may also be heard, and in bad cases a " plop " may be heard on rotation of the volume control under open-circuit-input, no-signal conditions. It is recommended that screened wire be used on the input to the first valve grid.

The design of the commercial models is quite simple, as the sketch shows. The left and right channels are separated from each other in respect of valves and tag-boards. The channels do however, have physical proximity, which is unavoidable in a compact design, i.e., at the input and output sockets, function switching, and ganged controls for tone and volume. The function switch used in both models is of the miniature push-button type; this merely satisfies the requirement of fashion and styling, and a multi-bank rotary switch would function just as satisfactorily, and is probably more simple to wire. At the end of the push-button assembly, a switch is fitted with heavy duty contacts for mains switching the complete installation.

Avoiding Hum Injection

To avoid unwanted hum injection into the adjacent switch, it is necessary to fit a copper foil screen between the mains and adjacent sections, and earth it. A separate earth lead should be run from the screen which should be insulated from the switch chassis to prevent hum loops. Wiring capacity feedback is greatly improved by an apparently untidy random wiring system, and any attempt at cabling of wires unless individually screened, other than earth returns, should be avoided.

Components of left and right channels or components far removed from each other, from a circuit point of view, should not be placed in close proximity for the same reason.

In the original circuit diagram shown in the October issue, the earth returns were shown separately connected to an earth chassis point. This is very important to avoid the so-called hum loops,

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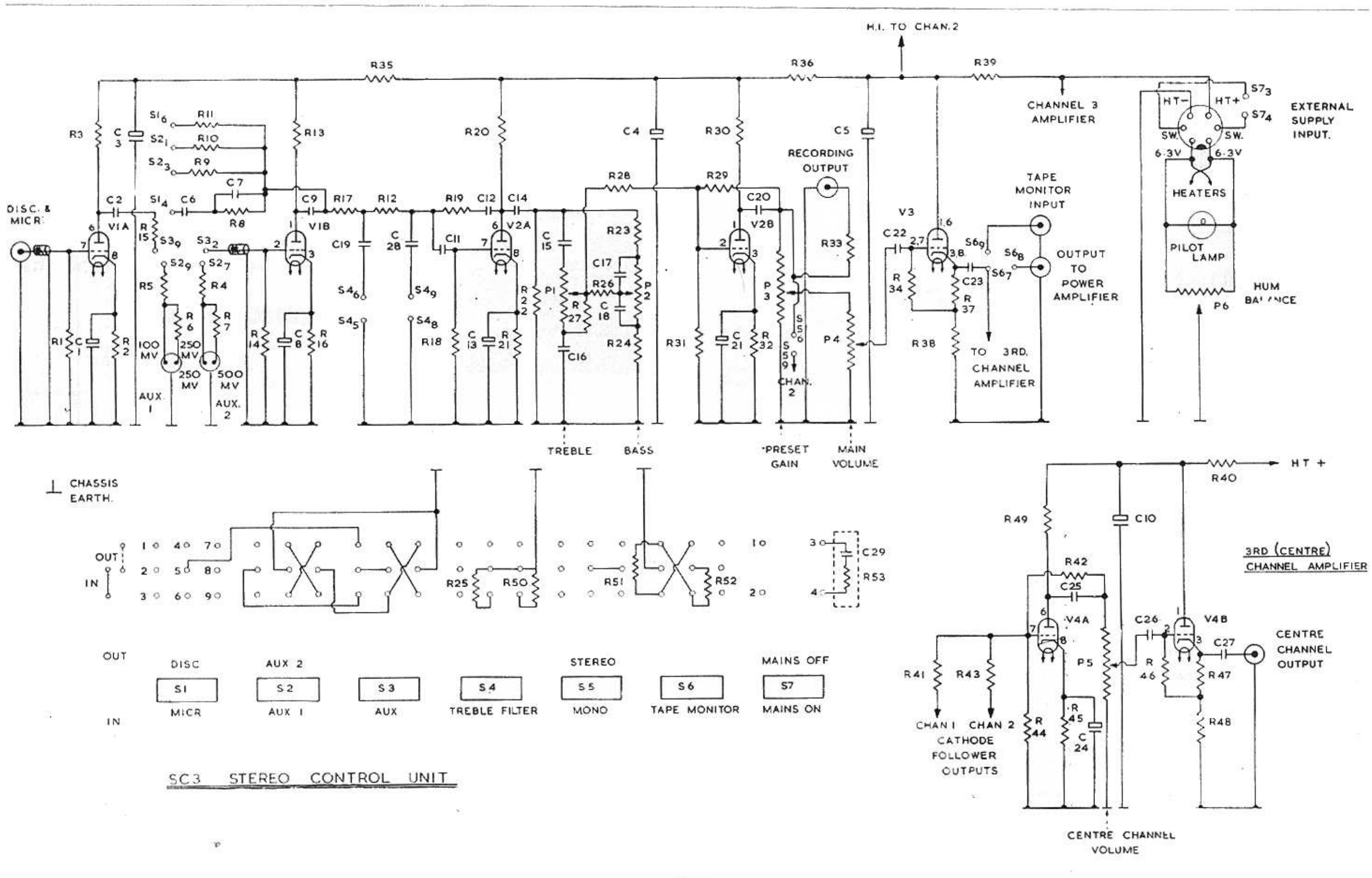


which are due to common impedance coupling, and which are the main causes of hum in amateur and some commercially built pre-amplifiers. Common impedance signal coupling can also produce instability by positive feedback. If the signal current of a late stage in the amplifier passes through a wire which is also carrying signal current to an early stage, then current feedback will take place, which may be positive or negative. Hum loops are usually the result of ripple current from an electrolytic smoothing capacitor being passed through the same wire as a signal current; hum is then induced in the circuit in proportion to the ratio of hum to signal currents.

Bus-bars Banned

It is therefore inadvisable to use bus-bar techniques of wiring, however advantageous it may appear from the point of view of simplicity in wiring. Similarly, connection should not be made to more than one point on the chassis, to prevent both signal and hum currents flowing in the chassis. Electrolytic capacitors should not be of the " earthy " can type, and should be insulated from the chassis. It is advantageous if the constructor thoroughly understands this principle of wiring if a highly stable, low hum and noise pre-amplifier is desired. **Fig. 1** shows how signal current feedback takes place in a common wire, and how hum currents from smoothing capacitors are fed to grids of amplifying stages.

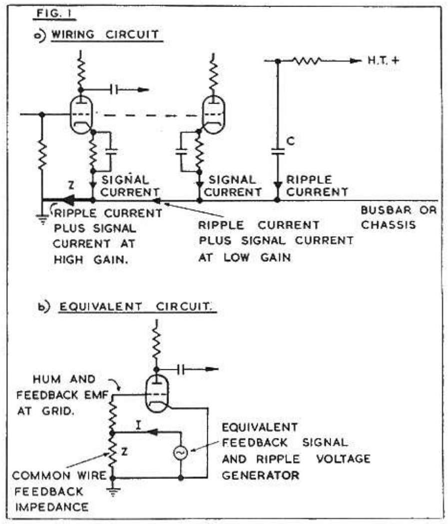
Concerning heater hum balance, it has been found that the best balance is obtained if all valves are wired the same with respect to pin numbers. That is, pin 9 should be wired to one side of the heater and pins 4 and 5 to the other side. Two-colour flexible wires assist in this respect.

Use Good Components

In addition to poor maintenance prospects, inferior materials and components can seriously mar potential performance characteristics; noisy and unstable resistors, leaking capacitors, and noisy controls and switches are useless. Resistors used in anode circuits should be of the metal-oxide type, and all others of the high stability carbon type. Capacitors should be polystyrene, polyester or mylar. The hum potentiometer is of the wirewound type, and all other potentiometers, moulded track carbon. This type is superior to inexpensive stud types and takes up far less room. The only problem from the constructors point of view, is that this may be difficult to obtain in small quantities. Component mounting boards should be made from grade "A" material, not the cheap commercial grade used in radio sets.

These circuits are not critical with effect to lay-out, provided the principles already outlined are observed. The sketch diagram shows how the commercial models are produced, but it would probably be advisable for the home constructor to provide a little more room to ease wiring and assembly. In the layout illustrated all the components connected with the circuit wiring are fitted to a pin type tag-board running the length of the chassis, and divided in right- and left-hand channels. The components concerned with tone control and equalisation are mounted to a separate board

How signal and hum ripple currents can be fed back is shown at (a), with the equivalent AC circuit shown at (b). Z is a common impedance in a feedback circuit



which is spaced on pillars from the front chassis behind the controls, and sub-assembly construction is therefore possible. The rear part of the chassis contains the valveholders, electrolytic capacitors, and all input sockets. The front chassis contains all the components for panel control, i.e., potentiometers and switches. The tag-boards are spaced from each chassis on hexagon pillars, and after they are wired in their respective sections, the two sections can be joined together, also by hexagon spacers.

Most of the circuit details have been dealt with in the previous articles; one point not mentioned is that the basic sensitivity of the SC3 unit may be increased, if desired, by reducing the feedback on

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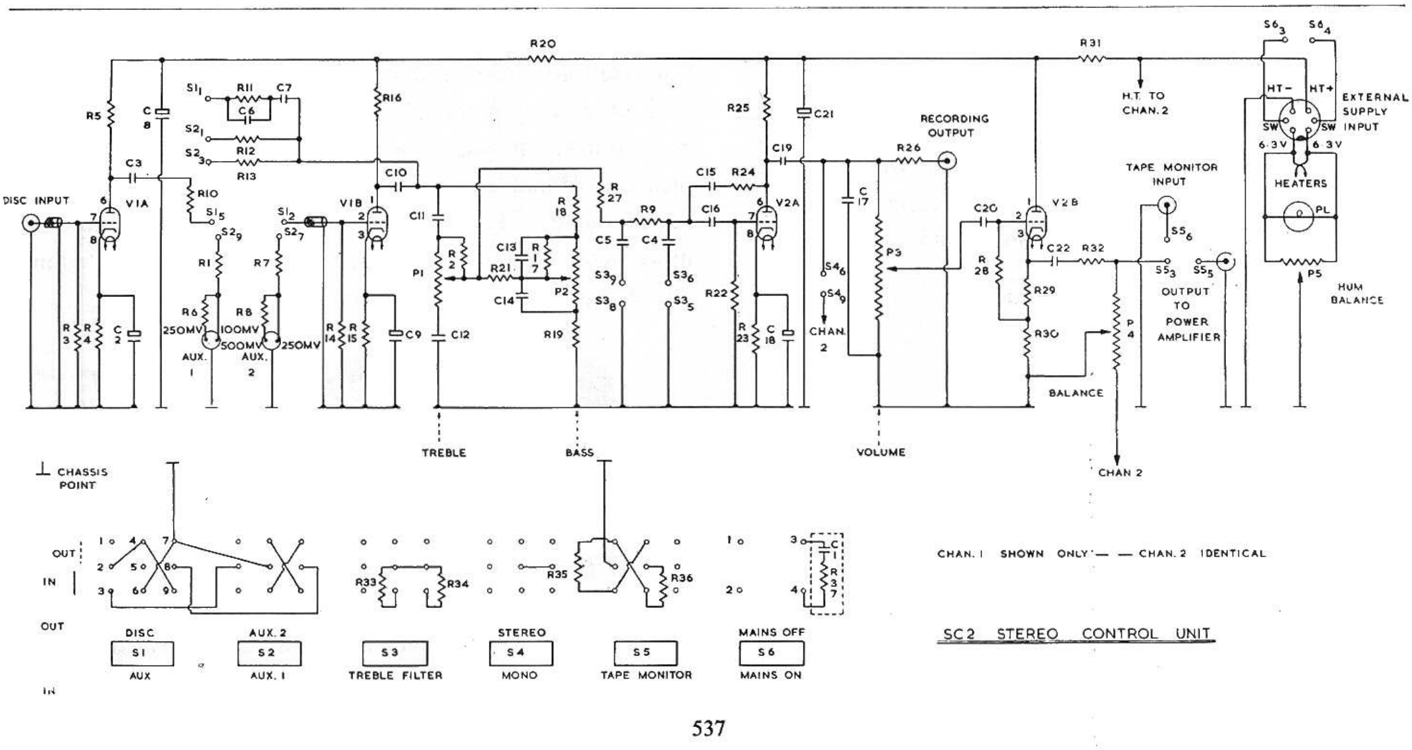
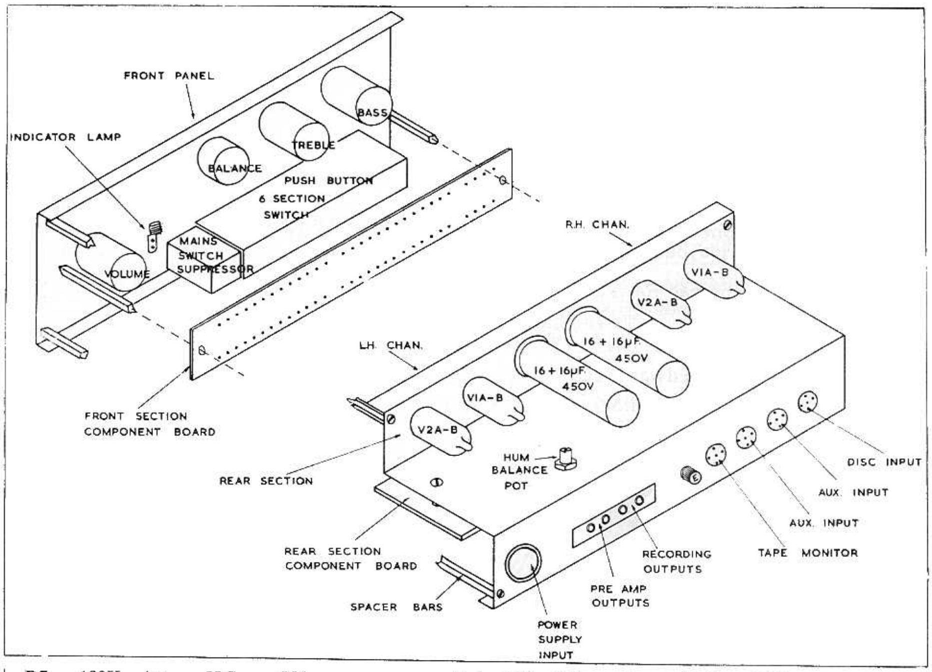


V2b. The pre-amplifiers are designed to give 500 mVs output from the input voltages specified; it may, however, be required to feed an amplifier having an input sensitivity of between 1 and 2 volts. R29 controls the feedback fraction, and can be adjusted to give the pre-amplifier the basic sensitivity required for insensitive power amplifiers.

A Third Channel

The SC3 has a third channel amplifier, V4a and V4b; V4a being the summation amplifier, and V4b cathode follower output. There is no reason, however, why this third channel circuit cannot be used with the SC2 circuit, if desired. This summation circuit should be used only with cathode follower output stages, as if fed from high impedance generators, high channel-to-channel cross-talk is likely.

A derived third channel system is thought by some to be a " gimmick " for filling in " the hole in the middle " on a poor stereo system. This arose from early experiments in summation with special transformers, which would permit the left and right channel amplifiers to feed a third centre channel loudspeaker. No control of the centre channel is available with such a system, which has proved to be quite unsatisfactory. The electronic system using an additional amplifier and loudspeaker is able to provide a wide sound stage with a flat wavefront, if the recording system or transmission method permits the recovery of a fully compatible monophonic signal, such as in stereophonic *broadcasting*. One of the requirements for stereo broadcasting is that a properly phased coincident microphone is used, giving a true monophonic signal when the outputs are commoned to provide a centre channel. The physics of stereophony, with particular reference to three channel working and the systems of recording and transmission which make this possible, will be described in some further articles in a few months' time.



PARTS LIST—SC 2 Control Unit

Resistors

R1	100K	½w	H.S.	5%	2
R2	150K	½w	H.S.	5%	2
R3	68K	½w	H.S.	5%	2
R4	3·9K	½w	H.S.	5%	2
R5	270K	Metal Oxide		5%	2
R6	180K	½w	H.S.	5%	2
R7	100K	½w	H.S.	5%	2
R8	100K	½w	H.S.	5%	2
R9	33K	½w	H.S.	5%	2
R10	100K	½w	H.S.	5%	2
R11	390K	½w	H.S.	5%	2
R12	330K	½w	H.S.	5%	2
R13	820K	½w	H.S.	5%	2
R14	680K	½w	H.S.	5%	2
R15	3·9K	½w	H.S.	5%	2
R16	220K	Metal Oxide		5%	2
R17	100K	½w	H.S.	5%	2
R18	47K	½w	H.S.	5%	2
R19	15K	½w	H.S.	5%	2
R20	10K	½w	H.S.	5%	2
R21	39K	½w	H.S.	5%	2
R22	100K	½w	H.S.	5%	2
R23	3·9K	½w	H.S.	5%	2
R24	680K	½w	H.S.	5%	2
R25	220K	Metal Oxide		5%	2
R26	100K	½w	H.S.	5%	2
R27	33K	½w	H.S.	5%	2
R28	1M	½w	H.S.	5%	2
R29	2·2K	½w	H.S.	5%	2
R30	47K	½w	H.S.	5%	2
R31	10K	½w	H.S.	5%	2
R32	4·7K	½w	H.S.	5%	2
R33	1M	1/10 w	Carbon	20%	2
R34	1M	1/10 w	Carbon	20%	2
R35	1M	1/10 w	Carbon	20%	2
R36	1M	1/10 w	Carbon	20%	2
R37	220Ω	½w	Carbon	20%	1

Capacitors

C1	0·05μf		1000v	Polyester or Mylar	1
C2	50μf		6v	Electrolytic	2
C3	0·047μf		400v	Polyester or Mylar	2
C4	1000pf	5%	400v	Polyester or Mylar	2
C5	1000pf	5%	400v	Polyester or Mylar	2
C6	200pf	5%	125v	Polystyrene	2
C7	910pf	5%	125v	Polystyrene	2
C8	16+16μf		450v	Electrolytic	1
C9	50μf		6v	Electrolytic	2
C10	0·022μf		400v	Polyester or Mylar	2
C11	560pf	5%	125v	Polystyrene	2
C12	3900pf	5%	125v	Polystyrene	2
C13	3900pf	5%	125v	Polystyrene	2
C14	0·022μf	5%	400v	Polyester or Mylar	2
C15	1800pf	5%	350v	Polystyrene	2
C16	0·01μf	5%	125v	Polyester or Mylar	2
C17	300pf	5%	125v	Polystyrene	2
C18	50μf		6v	Electrolytic	2
C19	4700pf	5%	400v	Polyester or Mylar	2
C20	0·01μf	20%	125v	Polyester or Mylar	2
C21	16+16μf		450v	Electrolytic	1
C22	0·1μf	20%	125v	Polyester or Mylar	2

Potentiometers

P1	250K	Log. Moulded Track–Double Gang	1
P2	250K	Log. Moulded Track–Double Gang	1
P3	250K	Log. Moulded Track–Double Gang	1
P4	100K	Lin. Moulded Track	1
P5	100Ω	Wirewound	1

Valves

ECC83	4

PARTS LIST—SC 3 Control Unit

Resistors

R1	68K	½w	H.S.	5%	2
R2	3·9K	½w	H.S.	5%	2
R3	270K	Metal Oxide		5%	2
R4	100K	½w	H.S.	5%	2
R5	100K	½w	H.S.	5%	2
R6	100K	½w	H.S.	5%	2
R7	180K	½w	H.S.	5%	2
R8	270K	½w	H.S.	5%	2
R9	510K	½w	H.S.	5%	2
R10	220K	½w	H.S.	5%	2
R11	270K	½w	H.S.	5%	2
R12	47K	½w	H.S.	5%	2
R13	220K	Metal Oxide		5%	2
R14	680K	½w	H.S.	5%	2
R15	100K	½w	H.S.	5%	2
R16	3·9K	½w	H.S.	5%	2
R17	47K	½w	H.S.	5%	2
R18	100K	½w	H.S.	5%	2
R19	270K	½w	H.S.	5%	2
R20	100K	Metal Oxide		5%	2
R21	2·2K	½w	H.S.	5%	2
R22	150K	½w	H.S.	5%	2
R23	68K	½w	H.S.	5%	2
R24	6·8K	½w	H.S.	5%	2
R25	1M	1/10 w	Carbon	20%	2
R26	39K	½w	H.S.	5%	2
R27	47K	½w	H.S.	5%	2
R28	100K	½w	H.S.	5%	2
R29	3M	½w	H.S.	5%	2
R30	100K	Metal Oxide		5%	2
R31	100K	½w	H.S.	5%	2
R32	2·2K	½w	H.S.	5%	2
R33	100K	½w	H.S.	5%	2
R34	1M	½w	H.S.	5%	2
R35	10K	½w	H.S.	5%	2
R36	10K	½w	H.S.	5%	2
R37	2·2K	½w	H.S.	5%	2
R38	47K	½w	H.S.	5%	2
R39	10K	½w	H.S.	5%	1
R40	10K	½w	H.S.	5%	1
R41	100K	½w	H.S.	5%	1
R42	120K	½w	H.S.	5%	1
R43	100K	½w	H.S.	5%	1
R44	100K	½w	H.S.	5%	1
R45	3·9K	½w	H.S.	5%	1
R46	1M	½w	H.S.	5%	1
R47	2·2K	½w	H.S.	5%	1
R48	47K	½w	H.S	5%	1
R49	220K	Metal Oxide		5%	1
R50	1M	1/10 w	Carbon	20%	2
R51	1M	1/10 w	Carbon	20%	2
R52	1M	1/10 w	Carbon	20%	2
R53	220Ω	½w	Carbon	20%	1

Capacitors

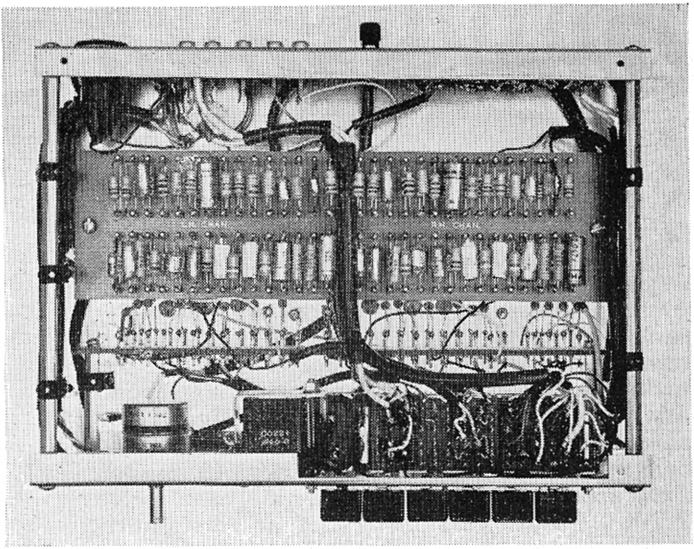
C1	50μf	6v	Electrolytic		2
C2	0·047μf	400v	20%	Polyester or Mylar	2
C6	1200pf	125v	5%	Polystyrene	2
C7	270pf	125v	5%	Polystyrene	2
C8	50μf	6v	Electrolytic		2
C9	0·022μf	400v	5%	Polyester or Mylar	2
C11	6800pf	125v	5%	Polyester or Mylar	2
C12	3600pf	350v	5%	Polystyrene	2
C13	50μf	6v	Electrolytic		2
C14	0·01μf	400v	5%	Polyester or Mylar	2
C15	560pf	125v	5%	Polystyrene	2
C16	8200pf	125v	5%	Polystyrene	2
C17	2200pf	400v	5%	Polystyrene	2
C18	0·022μf	400v	5%	Polyester or Mylar	2
C19	610pf	125v	5%	Polystyrene	2
C20	0·01μf	400v	20%	Polyester or Mylar	2
C21	50μf	6v	Electrolytic		2
C22	0·01μf	400v	20%	Polyester or Mylar	2
C23	0·1μf	125v	20%	Polyester or Mylar	2
C24	50μf	6v	Electrolytic		1
C25	0·01μf	400v	20%	Polyester or Mylar	1
C26	0·01μf	400v	20%	Polyester or Mylar	1
C27	0·1μf	125v	20%	Polyester or Mylar	1
C28	2700pf	125v	5%	Polystyrene	2
C29	0·05μf	1000v	Polyester or Mylar		1

Potentiometers

P1	250K	Log. Moulded Track–Double Gang	1
P2	250K	Log. Moulded Track–Double Gang	1
P3	1M	Lin. Moulded Track	2
P4	250K	Log. Moulded Track–Double Gang	1
P5	250K	Log. Moulded Track	1
P6	100Ω	Lin. Wirewound	1

Valves

ECC83	6



Radford Preamplifier Components

THE concluding article on the SC2 and SC3 control units for the M.A. 15 amplifier, published in last month's issue, gave detailed components lists for the use of home constructors. However, for those who have any difficulty the firm of Radford Electronics Ltd., Ashton Vale Estate, Bristol, 3, will supply sets of components. Particular items which may be difficult to obtain through the normal retail channels are the metal oxide resistors, push button switch unit and precision potentiometers. Readers are invited to write to Radford Electronics for details.

Pressure of space in the January issue prevented us from including a photograph of the Control unit. This is now shown on this page to give intending constructors some idea of the preferred layout.

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