THE

PARTRIDGE

MANUAL

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THE PARTRIDGE MANUAL

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FOREWORD

The business of this Company is the making and selling of transformers and chokes, components that are to serve their purpose to the greatest advantage. The desire to see the product fulfilling its function to the highest degree induced the founder, the late Dr. Norman Partridge, to write two booklets, " The P.A. Manual " and " The Partridge Amplifier Circuits," both of which had and continue to have a wide appeal. As both are out of print and the war years have intervened, with technical advance and changed outlook, this new " Partridge Manual " is introduced to replace the two.

The faithful reproduction of sound is a subject of particular and passionate interest for many readers of these manuals, and it is of concern to this Company that their efforts to expand the available techniques to encompass the full range of possible sensation, be assisted in every way possible. True high fidelity should be indistinguishable from the original, and as this target of perfection is approached it is inevitable that the newly developed methods and equipment will appear cumbersome and expensive in relation to the improvement gained. However, this is the way of advancement and simplification and cheaper and more efficient methods always follow. With this encouragement the chapters of this manual on this subject are presented.

Professional sound engineers might find of interest and use the chapters on P.A. and sound reinforcing, together with some of the circuits and design tables.

October, 1947

CONTENTS

SOUND AND HEARING

Nature of Sound.—Before considering the problems of electrical equipment it will be as well to review briefly the nature of sound.

Sound is a form of energy radiated from a vibrating body through the surrounding material medium as waves of pressure. These pressure waves result from the particles of the medium having inertia together with some form of elastic linkage one with another. The particles move to and fro in the direction in which the wave is travelling, in a manner similar to the movement produced in a train of goods wagons when a locomotive clouts the end one.

Harmonic pressure waves have the physical properties of frequency, intensity and waveform, and are steady or sustained. Changes of intensity, frequency or waveform introduce components which are transient or non-harmonic. Certain pressure waves affect the human ear, resulting in a sensation being registered and are known as sound or noise (noise being sound which is undesired by the recipient). Pressure waves outside the sensibility of the human auditory system are known by other terms such as supersonic waves, etc.

The Human Auditory System.—The auditory system records the physical properties of sound as sensations, viz., the sensation of pitch for frequency, of loudness for intensity, and of timbre for waveform. The human ear is a living, pressure sensitive microphone which acts as a Fourier analyser of the sound before transmitting the information to the brain. This property of the ear was formulated into an acoustic law by Ohm, and explains how relative phase changes of the components of harmonic sound are imperceptible. For transient sounds however relative phase changes can be sufficiently large so as to become a perceptible time interval and so altering the resulting sensation.

These facts are accounted for in a simple way by the assumption, which seems to be permissible anatomically also, that there are in the ear a large number of resonators each tuned for a different frequency. This mechanism explains how the Fourier analysis is carried out, and also makes it possible to suppose that the stimuli sent to the brain from each resonator depend on intensity and not upon phase.

Loudness and Intensity.—The auditory system judged as a scientific instrument is decidedly non-linear. Loudness (the subjective quality) is related to sound intensity (the objective quality) according to Fig. 1, which is taken for an average ear. They show the relationship between intensity levels and frequency for equal loudness. These curves were first published by Harvey Fletcher and are known as Fletcher curves. Sound intensities below the lowest

curve are inaudible, and those above the highest curve do not register as being any louder but register a sensation of feeling. These two curves are known respectively as the Threshold of Audibility and the Threshold of Feeling. Sound intensity is measured as decibels above a datum which is an r.m.s. sound pressure of 0.000204 dynes per square cm. at 20°C. and 76 cm. mercury, and which is approximately the threshold pressure at 1,000 c.p.s. for the average normal ear. The loudness of a sound is measured in phons the number of which is the same as the decibels of sound intensity at 1,000 c.p.s. judged by a normal observer to be as loud.

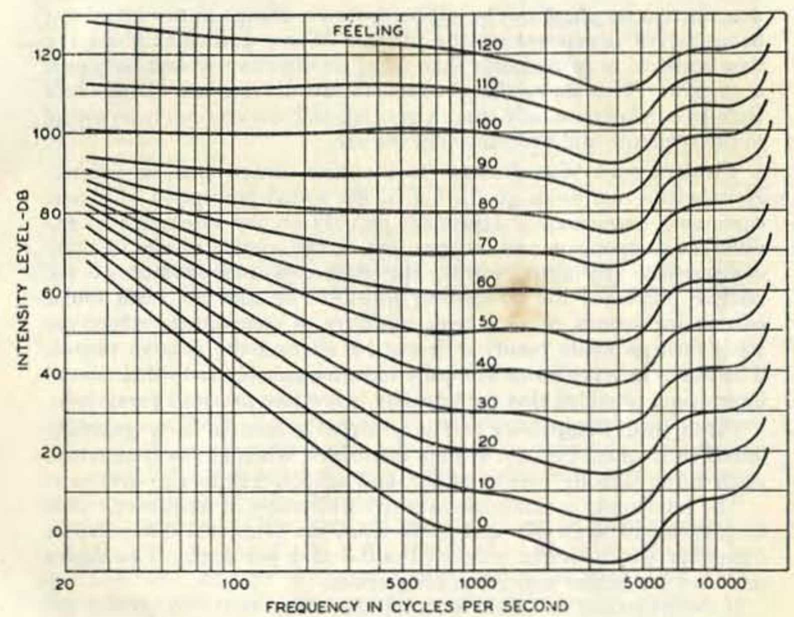


Fig. 1. Fletcher Curves.

From the irregular shape of these curves it will be seen that the ear will distort the loudness sensations of reproduced sound unless the intensity of that sound is the same as that of the original. This means, of course, the intensity of sound impinging on pairs of ears must be the same, and not that the loudspeaker must radiate the same sound energy as the original sound generator. Where this condition is not possible, or thought desirable, various forms of correction linked with the volume control are introduced. Examination of Fig. 1 will show how only very approximately this can be effected.

Non-Linear Distortion in the Ear.—Non-linear distortion in the ear is manifested in the fact that when a pure tone of sufficient intensity is heard, higher harmonics are formed in the ear (the octavo, the duodecimo, etc., are heard) and when two tones of different frequency are heard at the same time, new tones are formed which are linear combinations of the frequencies of the two tones (combination tones). The most obvious combination tone is the one of difference frequency, and this was the one first discovered. One speaks of objective and subjective tones according as the tones perceived are present or absent in the sound outside the ear. The inter-play of objective and subjective tones affect the perceived sensation. When a loud pure tone of about 100 phons is heard, subjective pure tones are produced. If about 10 per cent. of objective second harmonic is added in such a manner that its phase can be adjusted, then a change in the sound can be perceived dependent on the phase. At one particular phase the tone registers as being purer than when no objective second harmonic is present. This shows that the acoustic law of Ohm and Helmholtz's Rule (see below) are only true in general, and not absolutely so owing to the structure and mechanism of the ear.

Timbre and Waveform.—It has been found that in practice Helmholtz's rule holds good, that is, the sound perception of a tone containing harmonics is dependent entirely on the intensities of the different components and independent of the relative phases of these components. In other words, the timbre is independent of the relative phases of the component tones. The rule has been found correct for sounds of very large numbers of components where the wave form is made totally different by altering the relative phases. The different wave forms are quite indistinguishable as to their sound impression, provided that the intensity is low (see previous paragraph).

Pitch and Frequency.—The pitch of a note is fairly generally linked to frequency except at high intensities, when at low frequencies pitch varies directly with intensity, and at high frequencies inversely.

The human ear is extremely sensitive to changes of frequency. For frequencies between 500 and 4,000 c.p.s. the fractional difference in frequency which can be perceived is 0.3 of 1 per cent. This figure increases for higher and lower frequencies.

If the frequency of a note is rapidly fluctuated above and below the fundamental value up to about seven times per second then the ear can register these changes. Singers and players often do this and the effect is known as vibrato. If, however, the rate is increased above about seven per second, the ear loses the ability to follow the changes and registers the result as an amplitude variation with a complex mixture of harmonics. This phenomenon has a bearing on loudspeaker design, design of a single unit reproducing the full spectrum. Considering the example of such a speaker reproducing a treble note simultaneously with a bass note, the source of the treble note moves forward and backward with respect to the observer at a frequency equal to that of the bass note. Thus it can be understood how the treble note becomes frequency modulated by the bass note through this form of Doppler effect.

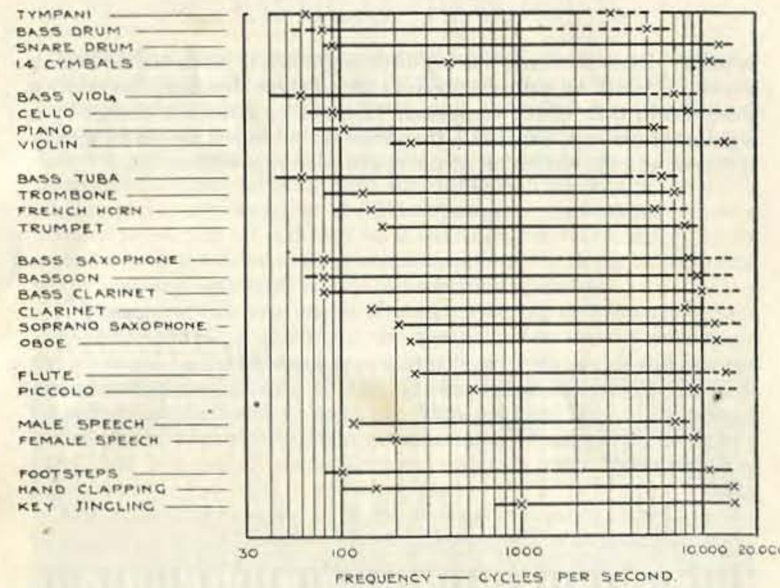


Fig. 2. Audible frequency range of various sound sources. The crosses mark the cut-off frequencies just detectable by the average person.

Range of Pitch.—Reference to Fletcher's curves show that there is a sharp fall in sensitivity in the average normal ear at about 15,000 cycles per second. This upper frequency limit of audibility becomes lower with increasing age of the observer.

At the lower end of the spectrum the sensitivity falls more gradually; between 50 c.p.s. and 25 c.p.s. the threshold of audibility changes by about 16 db. The most abrupt change occurs at around 18 c.p.s. where the sensation of hearing seems to change to one of feeling.

At bass frequencies the auditory system seems to provide the subjective observation of a fundamental note when presented with a series of harmonics. For this reason it is more difficult to distinguish the absence of fundamental when a sound reproducing system is made sharply to cut off below, say, 50 c.p.s. the reproduced frequencies then include the second harmonic of the lowest note commonly occurring in orchestral music.

Transients.—These components of sound form the consonants of speech and give character to music; the harmonic tones providing the colour and the transients the lines and outlines of the musical picture. Transients can be analysed into Fourier series like complex harmonic tones, and the components react on the ear like other sounds with the relative phases similarly having little significance with the following proviso. Since by nature transients are of short duration, the relative positions in time of the component frequencies must not change by a significant amount, otherwise the sensation produced would be totally different and the definition of the sound picture would suffer.

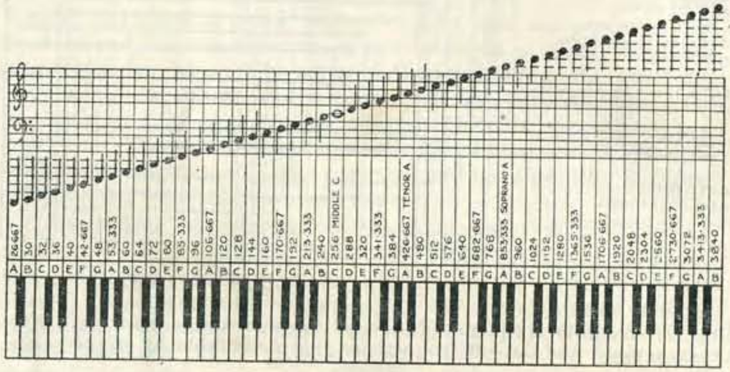


Fig. 3. Frequencies of piano notes. This can be of use in rough determination of resonant frequencies, approximate calibrations of oscillators, etc.

Masking.—When any sound is impressed upon the ear it reduces the sensibility of the ear to other sounds. This is known as masking and experiment has shown that a low tone will not obliterate to any degree a high one far removed in frequency except when the former is raised to a very high level of intensity and that a tone of higher frequency can obliterate a tone of lower intensity only when the two tones are close together. It follows then that the sensation produced by a complex sound is different in character as well as in intensity when the sound is increased or decreased in intensity without other distortion. In general, as the tone becomes more intense the low tones will become more prominent as the high tones are masked. Further change of character is produced by the generation of subjective tones as described earlier.

ACOUSTICAL PROBLEMS

General.—When a person of normal hearing listens to an orchestra in a large hall, the sound received by the ears is made up of that which has travelled the direct path or paths, and that which has travelled by a great number of different paths each including reflection from different solid surfaces. The relative proportions ~~of these~~ two components, of course, will vary with the distance of the hearer from the source, and it is found that the pleasantest effect is obtained when a definite proportion of reflected to direct sound is received. A room or hall which provides this optimum condition for the great majority of the listening occupants is said to have good acoustics.

A measure of this property is obtained from the reverberation time. This is the period required for the average sound energy density in the room or enclosure (and at a given frequency, the sound energy being initially in a steady state), to decrease after the source has been cut off, to one millionth of its initial value, *i.e.*, by 60 db. The unit is the second. The reverberation time, however, does not tell the whole story, but is a useful practical figure especially when expressed as a curve showing the value at each frequency.

The reverberation time of a particular room is increased by increasing the area of hard or sound reflecting surfaces and decreasing the area of soft or sound absorbing surfaces. Selective surfaces and shaping of the room are employed to produce the best curve of reverberation time/frequency. For example, an auditorium may be poor because of lack of damping for low frequencies only. Acoustic treatment which produces absorption for the high frequencies in such a room would make it worse rather than improve it.

Location of Microphone and Balance Technique.—A problem of practical importance to the acoustics engineer is that of the best location for a microphone and whether more than one microphone should be employed. The solution will depend on the directional properties of the particular microphones available. Most microphones are markedly directional for treble frequencies but not so for bass frequencies, with the result that if such microphones are placed too far away from the source the resulting quality is bass heavy. This is because the proportion of reflected sound to direct sound is increased with increased distance from the source, and if the microphone has marked directional properties in the treble only, then only the direct treble will be received. The ribbon microphone differs from other microphones in several respects. It is sensitive at the back as well as the front, and within an angle of 100° or so at both back and front the response is uniform. At the sides the response falls rapidly to dead silence on the plane of the ribbon. As a result of these properties the amount of reflected or reverberated sound picked up by a ribbon microphone is only about one-third of that picked up by other types. In practice this allows the microphone to be placed about twice as far away for the same ratio of indirect to direct sound.

Again the mode of operation of ribbon microphone exaggerates the bass frequencies when the microphone is located less than 18 ins. or

24 ins. from the source. Its use at distances less than these results in a " boomy " quality.

The object of correct location of microphones and of the technique of balancing the various instruments or other sources is to enable the microphone to pick up a sound pattern as close to that which a pair of ears favourably placed in the auditorium would do. As one would expect, this is best obtained by balancing the orchestra for a listener in the hall and then employing a single microphone at the best distance from the source to give the most pleasing proportion of indirect to direct sound.

However, owing maybe to indifferent acoustics, the required effect cannot always be obtained by the use of a single microphone. Where more than one have to be employed the important condition to be observed is that no instrument or singer is picked up directly by more than one microphone.

Acoustics of the Listening Room and location of the Loudspeaker.—A single channel system consisting of an ideal microphone, ideal transmission apparatus and an ideal pair of headphones would give the listener all but the sensation of audio perspective. Listening with two ears in an auditorium gives this sensation through the ability of the aural system to interpret the phase differences between the sound received by the respective ears. The ability to sense the distribution in space of the various instruments of an orchestra is a contributing factor in the total sensation experienced by the listener. This factor cannot be obtained with the ideal system supposed above. Neglecting the images based on previous experience, the impression obtained will be of the orchestra located somewhere on the plane perpendicular to and bisecting the line joining the ears. Two separate channels, each connecting a microphone to a single ear piece, with the microphones spaced similarly to a pair of ears and with each microphone having the same directional properties as the corresponding ear, would result in the received sound being indistinguishable from the original and including the factor of sound perspective. However, unless the microphones could be arranged to move and the movement interlocked with the movement of the listening ears, as the head is turned, the effect would still fall that much short of perfection.

Going back to the ideal single channel system, the replacement of the ideal headphones by an ideal loudspeaker, immediately involves a new set of difficulties. The listening ears hear sound directly from the loudspeaker and indirectly from the reflections and reverberation of the listening room. In general, the effect would be that of the source of sound being in another room connected to the listening room by the aperture of the loudspeaker. The audio perspective would be very much missed in the case of large orchestras or groups of talkers, but not at all with solo instruments or single talkers. As an example, listening to the Brains Trust (with high fidelity equipment) does not bring the sense of the talkers being round a table but that of each one in turn swiftly and silently moving to that hole in the wall, saying his piece and equally swiftly and silently making way for the next.

By using more than one loudspeaker fed from the same channel and

suitably located, this effect could be modified. The general result, however, would be an inability to tell where from the sound is coming, which may not be always pleasant.

The use of two channels with loudspeakers arranged in their spacing to correspond with the associated microphones would contribute greatly to the realism of reproduction. There would be a marked lessening of the " next room " effect. The use of further channels would result in further improvement, but not so marked. The introduction of F.M. transmissions on two channels might provide the opportunity for the B.B.C and listeners to combine in some interesting experiments in this direction.

The above consideration of ideals may assist the reader to arrive at the best compromise with a single speaker on a single channel. As the vast majority of such speakers concentrate the treble sound within a comparatively small solid angle about the axis, the location should be such that this beam is not interrupted by intervening absorbent material. As these upper frequencies give the listener the sense of direction from which the sound is coming, then this consideration influences the choice of speaker location and often provides an argument against the combined set and speaker. Many otherwise blameless radiograms are spoilt through the loudspeaker being mounted too low down.

Use of two or more loudspeakers each reproducing a Complimentary portion of the audio spectrum.—Some of the more difficult problems associated with the high fidelity reproduction of sound arise fiom the very great band width of the audible tone range. This, extending over nine octaves, is probably the widest encountered in the applied science of electronics. It is the limiting factor in the design of all the associated equipment, microphones, amplifiers, radio units, reproducers, recorders, etc., and the difficulties increase with increase of power, particularly so with loud speakers. Hence some high fidelity, high power installations overcome limitations by dividing the spectrum between two or more loudspeakers which, in fewer cases, are each fed from a separate amplifier. An arrangement commonly used in high-grade cinema installations consists of a bass speaker (" woofer ") of the moving coil direct radiation type mounted on a large flat baffle and a treble speaker (" tweeter ") of the moving coil exponential horn type.

Large diaphragms can be made which move with their entire surface substantially in phase up to a frequency of the order of one kilocycle per second, while small diaphragms behave similarly as pistons up to correspondingly higher frequencies. The latter, in particular, require some matching device such as an exponential horn for a reasonable transfer of energy to the surrounding air. An exponential horn behaves somewhat as a high pass filter having a cut off frequency dependent on the length of the horn ; the greater the size of the horn the lower the cut off frequency. A bass horn thus becomes a very large affair, and is usually not practicable through limitation of space unless some compact folded design is used.

The use of such multiple speakers enables a level frequency characteristic to be more easily obtained, reduces the effects of non-linear or wave form distortion occurring in the units themselves and, finally, reduces the frequency modulation of treble tones by the bass as previously described. The last two effects are the most important, due to the generation of combination tones.

Non-linear or waveform distortion as later defined is objectionable, not due to the resulting harmonics, but through the associated combination tones. Thus the Distortion Factor (as later defined) becomes merely a figure indicating the goodness or badness of a circuit with respect to these combination tones. The harmonics of the order usually met with in electrical sound apparatus being musically related merely alter the timbre of a particular sound. Since, in general, sounds of musical instruments or of the human voice are already very rich in harmonics, the proportions of which vary widely for different specimens of the same instrument as well as for the same instrument played differently, it is of little consequence that a relatively small percentage is added or taken away. Thus one notices how tolerably solo instruments are reproduced even on the meanest of equipment. In contrast, combinations of many instruments under similar conditions of reproduction lose clarity, and bear very little resemblance to the original. The individual instruments lose their distinction and become merged one with another due to the wide spectrum of sum and difference tones having very little musical relationship. Sound has become a noise.

It must be stressed also that the Distortion Factor is an adequate measure of the ability of apparatus to introduce combination tones when it is known for all relevant frequencies.

To obtain the full advantage it is obvious that energy must be fed to each loudspeaker of a combination only within the respective operative frequency range. Circuits which achieve this are known as dividing networks. Such networks are commonly connected between the output of an amplifier and the loudspeakers, but further advantage can be gained by insertion in an earlier part of the chain. This increases the expense, and usually it is not worth while dividing before the output valves. It is often of great value to divide between the output valves and the output transformer. Such an arrangement is given later in this manual. This circuit in addition to reducing the effect of loudspeaker distortion, reduces the Distortion Factor of the output transformer, imposes a better load on the valves at the extremes of the frequency band and enables loudspeakers of differing impedances and efficiencies to be used.

ELECTRICAL EQUIPMENT

Overall Specification.—Before considering in detail the various items of apparatus comprising the electrical equipment linking the broadcast or recording studio with the loudspeaker it might be worth while to write down the overall requirements for the highest fidelity reproduction. These are :—

(*a*) A frequency response characteristic, defined as the ratio of the sound power at any frequency usefully emanating from the loudspeaker to the sound power suppled to the microphone, to be flat within $\pm$ 1 db. from 30 to 15,000 cycles per second.

(*b*) The Distortion Factor, defined as

$$\frac{\text{Sum of squares of amplitudes of harmonics}}{\text{Square of amplitude of fundamental}} \times 100 \text{ per cent.},$$

to be less than 0.5 per cent. for middle and high frequencies and less than 2 per cent. for frequencies below 100 cycles per second, all at maximum modulation, and not to increase with reduced modulation.

(*c*) The time taken by a component of any frequency to pass through the system to be not more than a fraction of a millisecond or so different from that taken by a component of any other frequency. (This condition is peculiar to the satisfactory reproduction of transients).

(*d*) The phase change must be a minimum consistent with the frequency characteristic.

(*e*) The ratio of peak power at maximum modulation to unwanted noise expressed as decibels to be not less than 60.

(*f*) The power ratio of maximum to minimum modulation to be at least 40 db. and preferably 50 db.

Of the above characteristics (*a*), (*b*), (*c*) and (*d*) depend only on the economic factor for their achievement while (*e*) and (*f*), in the present state of the science, are mutually interdependent and cannot be described as anywhere near attainment. The ratio between orchestral fortissimo and pianissimo passages is of the order of ten million to one (70 db.) and if this ideal volume range is to be reproduced then a signal to noise ratio of about 80 to 90 db. will be required. With the amplitude form of modulation at present in general use a volume range of 25 db. is possible (from 5 per cent. to 95 per cent. modulation) and since frequency modulation can give an improvement in signal to noise ratio of about 30 db. then it is reasonable to expect a corresponding improvement in volume range. A later paragraph will discuss volume expanding networks which can be introduced at the receiving end.

In addition to the above listed electrical requirements, the equipment should possess the following features of the importance of which the operating engineer needs no reminding :

(*g*) It should be reliable and easy to service.

(*h*) It should be easy to set up and easy to operate.

(*i*) It should maintain its electrical characteristics throughout its service life.

(*j*) It should be of pleasing appearance.

(*k*) It should be as light in weight and as cheap in price consistent with all the other requirements.

Volume Expansion.—The background noise inherent in all known systems of radio or wire transmission and of recording, means that the volume range has to be restricted or compressed. This compression, as is well known, is carried out prior to transmission or recording and is achieved either by a hand-operated volume control or by electronic circuits doing the same function but without the ability to anticipate.

Ideally the compressor should be an amplifying device whose gain in controlled in inverse proportion to the peak programme envelope. As the programme envelope or peak volume is apt to vary as steeply and rapidly as the vibrations of a single note, the problem becomes a little difficult. The best practical scheme is one which reduces the gain rapidly in immediate anticipation of a loud passage and then acts more slowly to restore the gain to normal.

This compression being an alteration to the original is a form of distortion, and it is reasonable to attempt to correct for it at the receiving end by the use of a complementary expansion unit. Very satisfying results can be obtained provided that the law of expansion enjoys some nodding acquaintanceship with the law of compression and provided that the apparatus does not introduce distortion such as harmonic distortion of its own.

Essentially an expanding circuit consists of two portions, the one a valve amplifier whose gain is controlled by the other. A pair of push-pull variable-mu valves commonly comprise the first while the second consists of a condenser charged from the programme via a rectifier and having charging and discharging time constants in accordance with the desired expansion law. The d.c. voltage thus produced is used to control the gain of the first portion. The signal applied to the variable-mu valves must be sufficiently small so as not to result in appreciable harmonic distortion.

The development of such a circuit is a fruitful field of experimental research for the amateur.

Amplifiers.—Under this heading only audio frequency amplifiers will be considered since radio frequencv circuits of all kinds are beyond the scope of this manual. For information on these a reference should be made to such well-known text books as Langford Smith's " Radio Designer's Handbook " or Terman's " Radio Engineer's Handbook."

All amplifiers are, strictly speaking, power amplifiers, but the early stages of most amplifiers operate so as to develop maximum voltage and not necessarily maximum power.

Maximum power, irrespective of any other condition, is obtained when the load resistance is equal to the driving resistance. In an amplifier arranged for voltage amplification the load resistance (usually the external anode resistance of the valve in parallel with the grid impedance of the following stage) is made the largest value possible.

An inter-valve transformer enables the driving valve to work under conditions of maximum power and then a step up to the grid impedance to be obtained. Such a transformer also provides the facility for rapid clearing of temporary overloading of the grid circuit but has the disadvantages of being bulky and of introducing indeterminate phase changes outside the pass band making the application of feed-back more difficult.

Power amplifiers are referred to as Class A, Class AB_1 or Class AB_2, Class B_1 or Class B_2, or Class C according to the mode of operation of the valves. The last-named is only used in single frequency amplifiers where the harmonics produced can be filtered out by resonant circuits. Class A covers all stages of an audio frequency amplifier except some push pull output stages which for reason of economy are operated under Class B conditions. These conditions provide greater efficiency, by which is meant the ratio of audio power obtained, to the d.c. anode power supplied.

Class A Power Amplifiers.—The distinguishing feature of this mode of operation is the constancy of the anode current when the amplifier is operating. Substantially constant anode current flows all the time whatever the level of programme. Automatic bias by means of a cathode resistor can be used to advantage, and the regulation of the anode supply is not critical. The proper operation of class A requires a careful balance of load impedance, operating point and maximum signal voltage on the grid. Maximum power is obtained when the load is equal to the dynamic resistance of the valve or valves, but for maximum power at a given percentage of harmonic distortion the load must be greater in the case of a triode (usually two to three times) and less in the case of a pentode. In the latter case the load resistance is chosen so that the even harmonics are zero at or near the maximum swing, leaving the remaining distortion composed of odd number harmonics, mainly 3rd and 5th, and at outputs lower than maximum the even number harmonics increase and the odd numbers decrease in strength.

When operated in push pull class A both valves are driving at all times and over all portions of the cycle, therefore the anode to anode load is twice that required for a single valve. However, the value may be modified slightly to take advantage of the cancellation of the even number harmonics by the output transformer. The load is modified to reduce the odd number harmonics at the expense of the even order which then cancel out. This principle taken further is used in Class AB_1 or AB_2.

Class AB_1 and AB_2.—These modes of operation are used only in push pull for audio frequency amplification. The grid bias and a.c. grid swing are such that anode current flows for more than half but less than the entire electrical cycle. In AB_1 the grid swing is not sufficient to cause grid current to flow while in AB_2 it is so. The grid bias must be obtained from a battery or separate circuit, and the power pack must be designed to have good regulation, since the H.T. current varies with the programme level.

Class B_1 and B_2.—Here the grids are biased to cut off or, more precisely, to " projected cut off " which is the point on the grid bias axis which is cut by the projection of the straight portion of the anode current/grid bias curve. The suffix (1) or (2) indicate the absence or presence of grid current as described for AB. With Class B, however, it is almost invariably B_2 which is chosen.

As only one valve is in operation at any one time, its complement being at that instant biased to cut off, the load resistance to be referred into the anode of each valve is that required by one valve. This load resistance must appear between the anode and the centre tap of the transformer. The equivalent load appearing across the whole winding from anode to anode is thus four times that required for one valve. Most valve manufacturers quote this anode to anode figure. The value for AB is something in between that for Class A and that for Class B.

In addition to the ordinary valves there are valves specially designed for Class B operation. These have such a high slope characteristic that the projected cut off is approximately zero bias and are characterised by unusually small grid current with positive grid voltage. In addition the grid current characteristic is reasonably linear, thus easing the design of the driver stage.

Class B driver stage.—Because of the non-linear nature of the grid impedance of a Class B stage the driving impedance must be low ; the maximum value for a given distortion is usually quoted by the valve makers. Also the d.c. path across the grid circuit must be of low resistance for the easy flow of grid current. All this means that a transformer is necessary, usually stepping the impedance of the driver valve down.

The Partridge Control Circuit.—Modern high efficiency beam tubes require screen voltages maintained at a constant value irrespective of signal variations. The Partridge control circuit provides such a screen supply with very little waste of H.T. power.

The screens are supplied through an ordinary triode known as the control valve. The anode of this value is connected to the H.T. supply and the cathode to the screens. The grid of the valve is taken to a point in the circuit whose d.c. potential is the same as that at which it is desired to maintain the screens. If no convenient point in the H.T. supply to earlier stages is available then a high resistance potentiometer across the full H.T. can be used.

The control valve should be of adequate capacity to carry the required screen current, and should have the highest mutual conductance possible. The Mullard TT4 is a good type for most applications.

This circuit has the further advantage of smoothing and decoupling the screen supply to an extent dependent on the mutual conductance.

The patent rights of this most effective circuit are now owned and controlled by the Mullard Radio Valve Co., Ltd., to whom application should be made for permission to use.

This same principle of operation is used as the basis of design for many power pack circuits having perfect regulation.

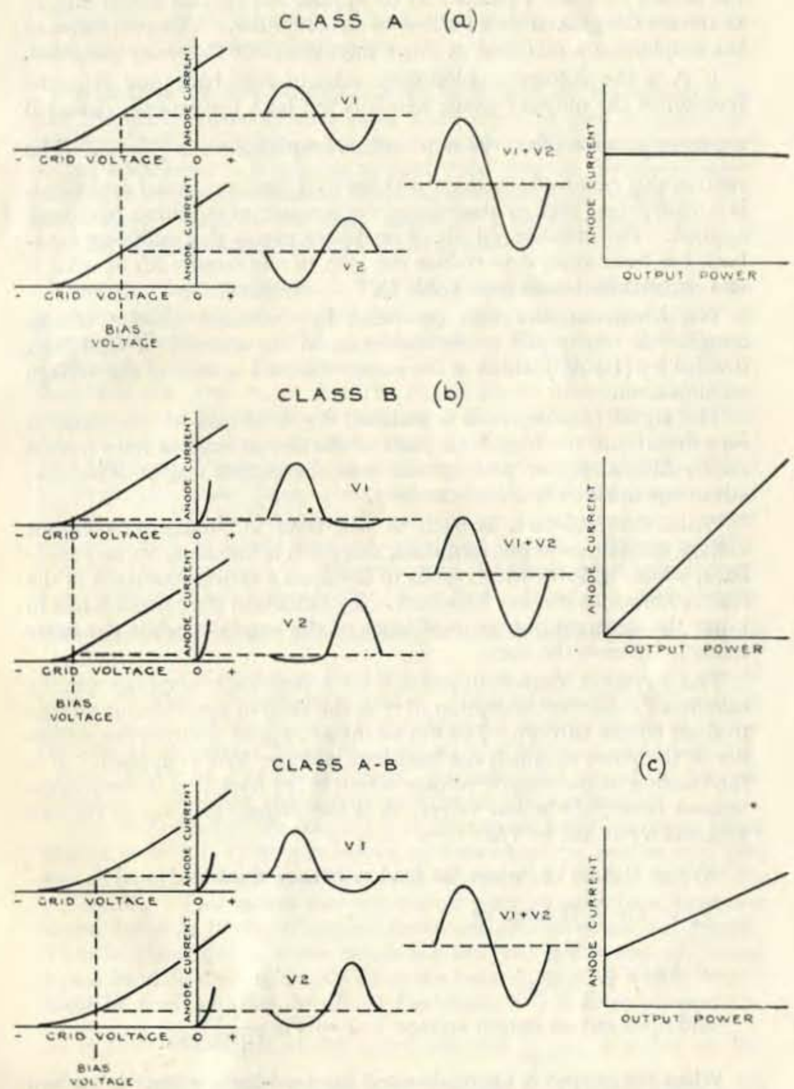


Fig. 4. Shows the operation of push-pull output valves working in Class A, Class B and Class AB. The transformer design and grid bias are factors governing the manner in which the valves function.

Negative Feed-Back.—Negative feed-back in an amplifier means that a voltage derived from the output is injected into an earlier part of the circuit in such a manner as to oppose the normal signal and so to reduce the gain of that portion of the amplifier. The properties of the amplifier are modified in other ways desirable for many purposes.

If A is the voltage amplification without feed-back and B is the fraction of the output-voltage which is fed back (in general, A and B are complex quantities) the new voltage amplification is $\frac{A}{1+AB}$. The ratio of this gain to A, the gain without feed-back expressed as decibels is a convenient way of describing the amount of negative feed-back applied. For instance, 40 db. of feed-back means that sufficient feed-back has been applied to reduce the gain to one hundredth of what it was without feed-back (see Table I).

The harmonic distortion produced by the same portion of the amplifier is now equal to the distortion in the absence of feed-back divided by $(1+AB)$ which is the same reduction as that of the voltage amplification.

The signal to noise ratio is reduced by feed-back if the noise is introduced into the high level parts of the circuit such as hum from a poorly filtered power pack getting into the output stage. This disadvantage must be watched carefully.

When the feed-back is such as will tend to produce a constant voltage across the output terminals, this form is known as voltage feed-back, while the form which tends to maintain a constant current in the load is known as current feed-back. By definition the former tends to lower the apparent output impedance of the amplifier while the latter tends to increase the same.

The apparent output impedance of a feed-back amplifier can be calculated from the conception of it as the ratio of open circuit voltage to short circuit current. Let V_{in} be the a.c. signal entering the amplifier at the point to which the feed-back voltage, V_{fb}, is applied. B is the fraction of the output voltage which is fed back. m is the magnification factor of the last valve. A is the voltage gain up to the last grid and V_g is $V_{in} - V_{fb}$

$V_{fb} = B.A.m.V_g$ when the load resistance is infinite (o.c.)

$V_g = V_{in} - B.A.m.\ V_g$

or $V_g = \frac{V_{in}}{1+B.A.m}$

and open circuit output voltage $V_{o.c} = V_{in}\frac{A.m}{1+B.A.m.}$

When the output is short-circuited the feed-back voltage is nil and $\therefore\ V_g = V_{in}$

and the short circuit current $I_{s.c.} = \frac{V_{in}\ A.m}{R}$ where R is the output impedance without feed-back.

$$\therefore\ \frac{V_{o.c.}}{I_{sc}} = R \text{ output} = \frac{A.m}{1+B.A.m}\frac{R}{A.m} = \frac{R}{1+B.A.m}$$

Thus the output impedance is reduced by voltage feed back in the same ratio as the reduction of gain.

Variations of Feed-back.—Feed-back can be manipulated to give frequency characteristics other than level, to give reduction of distortion without alteration to gain, or without alteration to apparent output impedance. It can be applied over links in the chain other than valve stages, for instance, from the rectified signal picked up by a local receiver back to the early stages of a transmitter, or from a recording head to the early stages of the amplifier.

Design considerations.—The design problems associated with negative feed-back are solved through the practical application of Nyquist's Regeneration Theory published in 1932 in the Bell System Technical Journal. For instance, if it is desired to apply 10 db. of feed-back over two stages then the phase change introduced by these two stages and the feed-back path must be less than 180° while the gain of the same loop remains within 10 db. of the mid band gain. At a frequency a little way beyond the one at which the gain has fallen by 10 db. the phase change can equal or exceed 180° and a more rapid fall in gain thus introduced. Where the loop contains only discreet elements such as resistances and capacities there is a unique relation between the frequency characteristic and the minimum phase change. For instance, when a capacity is fed with a.c. from a resistive source the phase of the current approaches $\frac{\pi}{2}$ and is in advance of the supply voltage; the voltage across the condenser falls at the maximum rate of 6 db. per octave of frequency change. This relationship between the phase change and the frequency characteristic fails to obtain (1) when the circuit includes a transmission line; (2) when at some high frequency the valves or transformers or chokes fail to act as lumped constants; (3) when the circuit includes an all-pass section.

The rate of fall of the frequency characteristic for a minimum phase change of 180° is 12 db per octave, and therefore the rate of loop gain cut-off in a feed-back amplifier must be less than 12 db., say, 10 db. per octave. This means that the cut-off interval must be at least one octave for each 10 db. of applied feed-back plus an octave for margin. Thus for the amplifier whose circuit is given later in this manual, having a pass band 30 c.p.s. to 15,000 c.p.s. the band width over which control must be exercised for 40 db. of feed-back is 5 octaves greater, *i.e.*, $32 \times 15 = 480$ kc.p.s. at the upper end and $\frac{30}{32} = 1$ c.p.s. at the other. The relation between loop gain and loop phase change can be used as the basis of design with a full consideration of all the elements which contribute to the loop gain characteristic.

Amplifier Construction.—Successful amplifier construction depends to a great extent on a flair for laying out the components to the best advantage. The mind must be accustomed to considering the various connections and circuit points in terms of :—

(*a*) the impedance from particular points to ground or chassis.

(*b*) the a.c. potential of particular points with respect to ground or chassis.

(*c*) the signal current carried by particular conductors.

(*d*) the phase relationship of the various points in the circuit.

Low impedance connections and some points of low a.c. potential are best wired in cable form while all others are wired point to point by the shortest sleeved wire. Wires carrying heavy a.c. must be twisted, go and return together. Components should be laid out in their correct respective positions in order of progression bearing in mind these wiring considerations. A good rule is so to space the components that any one can be taken out without having first to remove more than one other. Greater spacing must be given to parts giving off heat or heat reflecting metal screens can be provided to protect components such as electrolytic condensers. Inductance coils should be mutually so orientated that the minimum inter-coupling obtains. If the power supply is on the same chassis great care must be taken to separate the power components from the low level parts of the circuit. In this connection it is a great convenience to have the transformers and chokes fitted with mountings which enable the component to be fixed with its coil axis in any required orientation.

A final tip about wiring is never to allow the relatively high signal currents of the later stages in flowing through particular leads to build up even a minute voltage which is connected in any way in between the grid and cathode of the first stage. This rather obvious requirement is often overlooked when the chassis is rather carelessly used as the return lead for the signal currents of the various portions of the circuit.

In order to ensure stability in a high gain amplifier it is essential to keep the stray coupling between the output and input circuits small under all conditions. Since the input and output lines may frequently be run in close proximity, especially on outdoor equipment, it is necessary that both circuits be adequately screened and balanced.

Upon completion the wiring must be re-checked against the diagram and inspected for good soldered contacts. When this has been done to satisfaction the external circuits can be connected and the power supply turned on. Heater voltages should be checked at the valve pins. The output valves can then be inserted, and after seeing that the heaters are alight, the rectifier valve. If all is normal the remaining valves can be put in, in reverse order, checking all for feeds and making sure that the valve makers maximum voltages and currents are not exceeded.

Hum in Audio Frequency Equipment.—The most frequent causes for hum may be enumerated as follows :—

(*a*) From audio frequency action, which can be due to—

(1) insufficient filtering of the H.T. (50 c.p.s. or 100 c.p.s. according to whether a half-wave or full-wave rectifier is used) ;

(2) capacitative coupling from a source of hum to grids, inside or outside valves; this is reduced by reducing the grid impedance ;

(3) inductive coupling between transformer or choke coils;

(4) leakage between cathode and heater, owing either to poor insulation or to actual emission by the heater.

(*b*) From combined audio and radio frequency action. This may be the result of—

(1) hum modulating the carrier within the R.F. amplifier usually due to insufficient H.T. filtering;

(2) R.F. picked up from the mains. This is where the electrostatic screen in the mains transformer is useful;

(3) R.F. radiated by certain rectifiers mostly of the mercury vapour type. This is cured by shorting the electrodes to ground with small r.f. condensers.

Other Unwanted Noise.—Such noise arises from the following sources :—

1. Microphonic noises caused by the vibration of components usually in stages followed by high gain ; valves and variable condensers are the most frequent causes, and the trouble is often avoided by the correct choice of component or by designing the suitable anti-vibration mountings with or without acoustic screens.

2. Bad contacts in wiring, in screening or inside components such as batteries, condensers, etc. ; the mode of elimination is obvious.

3. Shot noise in valves followed by large magnification. The anode current of a valve is not absolutely steady, the electron stream rather resembles a crowd going to a football match, steady on the average but the individuals packed a little tighter here and a little slacker there. The effect is that of a steady d.c. with a random fluctuation superimposed. The frequencies of the fluctuations is distributed quite uniformly over the entire spectrum, and the sum total which is finally reproduced depends on the frequency band width. Valves working at very low levels must be very carefully chosen for low noise. The noise of a valve is frequently expressed as a resistance connected in the grid producing an equivalent thermal agitation noise (see below).

4. Thermal agitation or Johnson noise is due to the random movement of the free electrons in any conductor, and is proportional to the absolute temperature of that conductor. It is therefore advisable to keep low level circuits as cool as possible. At normal temperatures thermal agitation noise produced in a conductor of resistance R ohms and measured for a band width of $f_2 - f_1$ is given by $E = 12.6 \cdot 10^{-11} \sqrt{R(f_2 - f_1)}$ volts. This noise is produced in the series resistance part of any impedance, the reactive part does not contribute to the noise. Thus in the design of an amplifier to follow a ribbon microphone or moving coil pick-up an input transformer is of great value in reducing noise. A necessary voltage step-up is obtained with negligible increase of noise relative to signal since only the d.c. resistance of the windings contributes to that noise and the increase of signal obtained makes less important the shot noise of the first valve. The

greater the step up of the input transformer the better, but as the actual step-up obtained for a given band width depends on the core material, its size and the arrangement of windings, the importance of good design cannot be overstressed.

5. Site noise consisting of radiated atmospherics and man-made static exists everywhere, and is of importance only in radio frequency amplifiers and receivers. The only procedure is effective screening and the use of aerials which discriminate in favour of the wanted signal.

Screening.—Electric energy is conveyed from one part of a circuit to another by means of—

(*a*) An electric field;
(*b*) A magnetic field;
(*c*) Leads carrying current which in turn may produce electric or magnetic fields.

Electric fields are screened by sheets or wire mesh of conducting non-ferrous metals connected to frame or earth potential. For complete electric screening of a portion of a circuit the screen must take the form of a box completely enclosing that particular portion.

Magnetic fields at high frequencies are screened by means similar to those used for electric fields, the screening action being obtained through the reactive eddy currents produced in the conducting material. Very low frequencies such as 50 c.p.s. are completely unaffected by such screening and the only effective method of screening a component or portion of a circuit from such a field is to divert that field through material of high magnetic permability such as Permalloy or Mumetal. Sheet steel such as used for chassis making is practically useless at mains frequency. The screen thus takes the form of a box which will reduce the unwanted induction by 40 db. or more, provided the material is adequately thick and the shape and general arrangements are right.

Power Supply Units.—Units operating from 50 c.p.s. mains are of two broad classifications, one producing a supply which is independent of main's fluctuations and the other which is not. The former involves some form of electronic control, and reference should be made to the technical press for the many circuits that have been published.

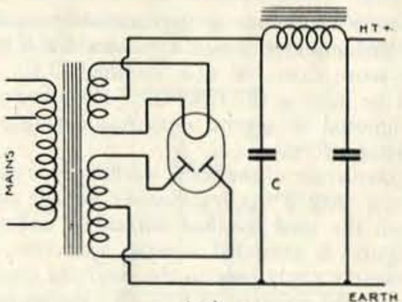
Of the other more usual kind, the design is based on the particular rectifier chosen and on the data supplied by the makers. The procedure is quite straightforward and to rule of thumb except the calculation of the necessary smoothing, towards which end the following information is useful.

With full wave rectification and a reservoir condenser, the ripple, at the fundamental frequency double that of the mains, developed across the reservoir condenser is given with sufficient accuracy by the formula $V_{r.m.s.} = \frac{150\,I}{2fC}$ where I is the d.c. in m.a., f the mains frequency and C the reservoir in mFds. which for a mains frequency of 50 c.p.s. becomes $V_{r.m.s.} = \frac{1.5\,I}{C}$.

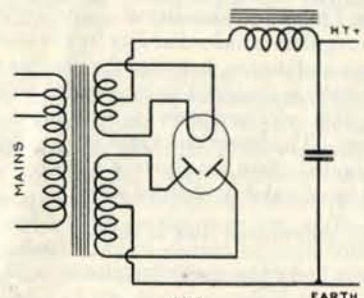
The second harmonic of this ripple is one-tenth or so of the fundamental and thus can be neglected. One stage of filtering consisting of a choke, reactance $2\,\pi.\,100\,L$, and condenser C_1, reactance $-\frac{10^6}{2\pi.\,100\,C_1}$, will reduce the ripple to approximately

$$V_{r.m.s.} = \frac{1.5\,I}{C}\;\frac{\frac{10^6}{2\pi\;100\,C_1}}{2\pi\;100\,L - \frac{10^6}{2\pi\;100\,C_1}} = \frac{1.5\,I}{C}\;\frac{10^6}{(2\pi\,100)^2\,LC_1 - 10^6}$$

Reservoir Condensers.—The choice of a suitable reservoir condenser is important, and generally speaking electrolytic condensers are not suitable. There are two reasons:—(1) The highest working voltage generally obtainable is of the order of 500v. The peak voltage to which the reservoir condenser is charged during each cycle is 1·4 times the transformer R.M.S. voltage. Hence the highest transformer voltage that can be employed with a 500v. electrolytic reservoir condenser is 350—0—350v. (500 divided by 1·4). (2) Electrolytic condensers cannot pass more than a very limited A.C. ripple current without rapid deterioration. In the case of most 500v. working types the maximum ripple is limited to 100 m.a. Now the ripple current



(a)



(b)

Fig. 5. (a) illustrates the usual condenser input to filter and (b) shows the choke input used where good H.T. regulation is required.

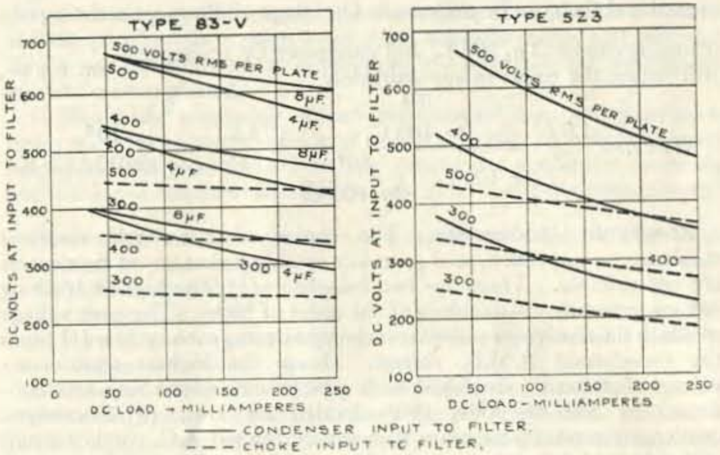


Fig. 6. Characteristic curves of the 83v (or 5V4) and the 5Z3 (or 5U4) rectifiers when operated in the circuits shown in Fig. 5.

passed by a reservoir condenser is approximately equal to the D.C. current taken by the amplifier, hence it follows that if an H.T. unit is required to pass more than 100 m.a. rectified D.C., an electrolytic condenser cannot be used as the reservoir. This important technical point has been ignored in several circuits emanating from sources expected to be better informed.

An electrolytic condenser of the 500v. working type can be used with safety as a reservoir only if the transformer voltage does not exceed 350—0—350v. and the total rectified current is under 100 m.a. If either of these figures is exceeded a paper condenser must be used. The preceding remarks apply only to the reservoir condenser because the ripple current in the condensers following the smoothing choke is negligible.

Rectifier circuit with choke input.—Full wave rectifier circuits can be improved in regulation and reliability by the inclusion of a choke, of certain inductance and design, between the rectifier and the reservoir condenser. This choke is designed to be of value somewhat above the lowest inductance that will maintain a continuous flow of current through the rectifier. The lower the value of the choke resistance the better the regulation, therefore the choke is chosen to be of inductance just above the minimum value as defined above.

The condition for continuous flow is that $\frac{2\pi f L}{R_{load}} > \frac{2}{3}$, where L is the choke inductance and f the mains frequency $\times$ 2.

The D.C. Voltage at the output is $0{\cdot}9\ E_{r.m.s.} - I\ (R_{rect} + R_{choke})$.

The ripple across C is $\frac{0{\cdot}6\ E_{r.m.s.}}{(2\pi f)^2\ L\ C - 1}$ where f is twice the mains frequency.

The input choke has to have certain specific properties over and above those of the ordinary smoothing choke. The winding must be capable of withstanding the full ripple voltage, and the larger switching surges. The mechanical clamping must ensure silent operation even with this large ripple voltage.

The Swinging Choke.—If the current taken by the load varies between two limits I_1 and I_2 the regulation of the circuit to this variation can be made almost perfect by arranging the choke to be near saturation at the lower limit I_1 and so at higher currents the inductance of the choke falls rapidly and at the upper limit the circuit is functioning almost as a capacity input one. Such a circuit needs careful design starting with the manufacturers' data on the rectifier.

Class B circuits require very carefully designed power pack units if distortion is to be kept within reasonable proportions. The good regulation provided by a swinging choke circuit is valuable to this end, so long as additional smoothing is provided. The final smoothing condenser has to be adequately large since it is an essential part of the Class B circuit (each valve working independently over its particular half cycle).

Tone Controls and Tone compensated Volume controls. — Audio amplifiers are frequently provided with controls for altering the bass and the treble characteristics. These are termed tone controls

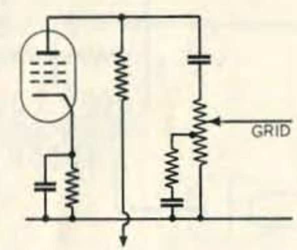


Fig. 7. Shows a method of adjusting the frequency characteristic of an amplifier to correspond with the gain.

and are for the purpose of modifying the character of the amplifier output to give new æsthetic effects, to compensate for acoustic characteristics, reduce hum, scratch, etc. Such circuits should modify the bass and treble without affecting the middle frequencies, although in many cases the ideal is not reached. For frequency characteristic slopes of up to 6 db. per octave circuits employing resistance shunted by capacity or capacity in series with resistance, are sufficient. For sharper characteristics, resonant circuits should be used with means of varying the effective Q of the circuit and of varying the L/C ratio.

The characteristics of the ear are such that the apparent loudness of bass tones relative to treble is less as the volume level is reduced. In order to correct for this, volume controls can be arranged so that as the gain is reduced, the reduction is less for bass than for other tones.

Such an arrangement is termed a tone compensated volume control and an example is given in Fig. 7.

P.A. work often calls for the reverse process where voice levels larger than life are needed. At such levels the bass chest tones become overpowering and speech unnatural with reduced articulation. To eliminate these effects the bass and to a much less degree the treble must be attenuated.

Summarising, P.A. apparatus being generally used to reproduce sounds at or above the natural intensities, requires adjustable controls for attenuating the bass and the treble. Radio or gramophone amplifiers for home use, normally operated at or below the original sound volume, need tone correcting circuits for boosting treble and bass. These conclusions do not take into account the suppression of needle scratch and side-band twitter, correction for recording characteristics, room acoustics, etc., each of which must be treated as a separate problem.

SOUND REINFORCEMENT AND PUBLIC ADDRESS

General.—Sound Reinforcement Equipment is designed to make the original source of sound more easily audible to a large audience. The criterion of success in the use of such equipment is that the listeners are not made conscious of its operation. Such success is achieved by the very careful choice of components, the microphones for their directional properties, amplifiers of adequate audio power output, loudspeakers suitably placed, and last but not least the design of the auditorium itself. Since in all cases the orchestra or speaker can be seen and heard by the audience, acoustical reaction between loudspeakers and microphones must be most carefully eliminated.

Public Address Equipment is used for direct entertainment (as from recordings), for general announcements or paging, for controlling and directing crowds, for adding to the interest of events, such as car racing, sports, etc., by means of a commentary, for publicity and electioneering, for use in place of expensive instruments such as a peal of bells or a church organ, for the provision of sound effects, etc., etc.

Low Level Circuits.—A gramophone pick-up, sound head, radio receiver unit, microphone or similar device provide the initial electrical audio frequency energy to be amplified. All except the radio receiver unit produce a low level signal which has to be treated with care so that background noise does not become unduly great. In the choice of component such as of a microphone the inherent background noise has to be considered, and it is in this respect that the carbon type is so bad.

When microphones and pick-ups are located some distance from the main amplifier then connection must be made via circuits of impedance neither so high as to pick up noise by capacity out-of-balance, nor so low as to do the same by magnetic induction. It is found that a circuit of impedance 100 to 600 ohms is a good compromise and provided the screening and balancing is good, it can be used for distances up to several hundred yards even with such insensitive devices as ribbon microphones or moving coil picks-ups. Two screened and balanced transformers are needed, one for stepping the microphone impedance to the line impedance and the second, usually part of the amplifier, for stepping the line impedance up to the grid of the first valve. The latter should give a step-up sufficient to provide a signal which is adequately high with respect to amplifier valve noise. To achieve this in the case of the most insensitive devices such as ribbon microphones means that the transformer has to be designed to the limit, and must take account of the input capacity and resistance of the valve. The cost of such a well designed component often appears high, but the economics have to be balanced against the technical result.

Adequate screening often demands mumetal boxes enclosing the transformers. Such boxes need careful designing, and it is not sufficient to specify the material from which they are to be made. It is better to state in decibels the reduction of pick-up of an external 50 c.p.s. field by the component when it is so enclosed.

The usual kind of screened microphone cable has a surge impedance of the order of 100 ohms which is suitable for the circuits described. It might be helpful to point out here that the surge impedance of a cable is not to be confused with the D.C. resistance of a particular piece but is the impedance of that cable to an electro-magnetic wave motion along it. Its value depends on the self capacity between wires, self-inductance, conductor resistance and insulation resistance, all per unit length of the cable. Only where the runs are long compared to the wavelength of the wave motion is it necessary to match to this surge impedance.

It is often found necessary to employ pre-amplifiers. This is the term for an amplifier located close to the microphone or pick-up so as to raise the signal level before transmission over a longish line having perhaps a high noise pick-up. Such amplifiers are usually single stage employing a low noise valve, having an input and an output transformer, and being designed to handle up to the maximum level likely in the particular circumstances without distortion. In some instances such as with condenser microphones the amplifier is built in with the instrument and power supplies have to be connected by a multi-core cable.

Mixing and Fading.—There are many ways of mixing input channels ; some are too crude to be satisfactory, and others too exact to be economical. The circuit chosen will depend on the sensitivity

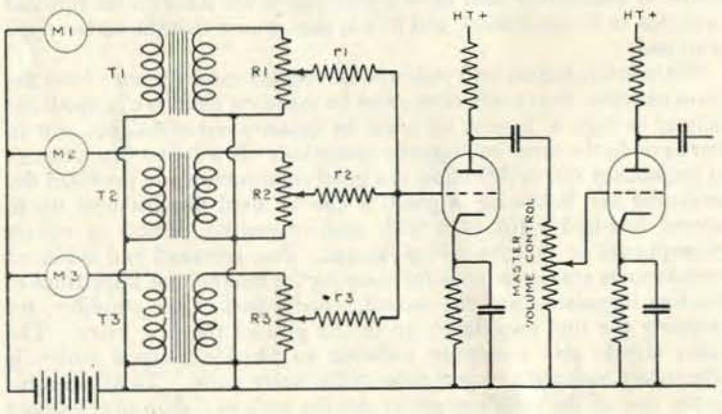


Fig. 8. Method of mixing three high sensitivity microphones—R_1, R_2, R_3, r_1, r_2 and r_3 are each 250,000 ohms.

of the microphones or pick-ups and the extent of the stray noise field in the vicinity of the equipment. The circuit used should be judged by its effect on the signal to noise ratio at all levels and on the general quality in other respects such as frequency characteristic.

Fader circuits should also be judged in the same manner.

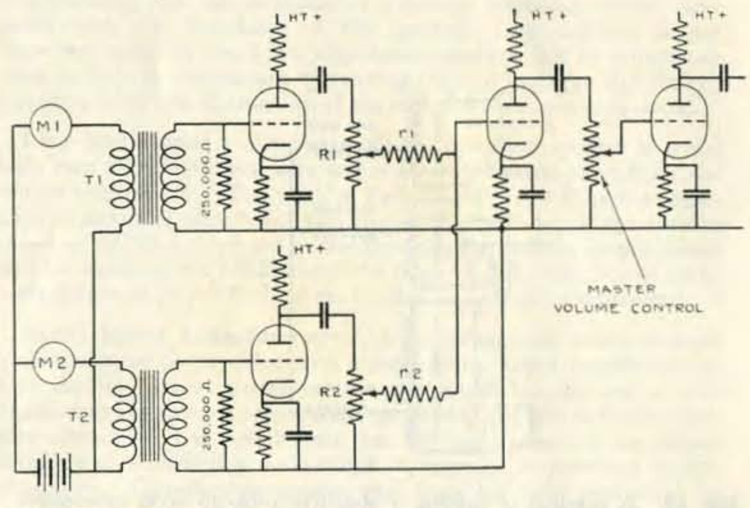


Fig. 9. This circuit is suitable for low sensitivity microphones. Extra channels can be added. R_1, R_2, r_1, r_2, etc., are 250,000 ohms.

Output Circuits — Varying Loads.—Many occasions arise in practice when it is not possible to operate under the ideal load conditions required by the output valves. In radio relay station, hotel and school installations and in the home where extension speakers are used, it is essential to control the volume of individual speakers and to switch speakers out of circuit without altering the ratio of the output transformer. Advantage can be taken of the fact that with triodes in Class A and also in certain Class A—B circuits the anode load can be increased (never reduced) above the optimum value providing the output power is reduced in the inverse proportion. Suppose four 15 ohm speakers are operating in parallel from a 12 watt output stage. The transformer should be wound for 3·75 ohms output. If two speakers are switched off, the load on the 3·75 ohm output will now be 7·5 ohm (15 divided by 2). However, if no alteration be made to the setting of the amplifier volume control, the power supplied to the remaining speakers will not increase perceptibly and the output power will therefore drop to one-half. Under these conditions no distortion will occur, but if the amplifier volume control is turned up in an attempt to feed the full 12 watts into the mismatched loud harmonic distortion will at once become apparent. Pentodes and many Class B circuits are quite unsuitable for conditions such as the preceding in which the load impedance is varied.

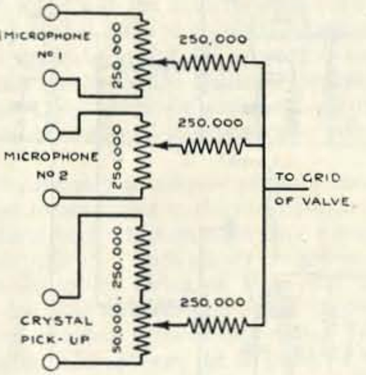


Fig. 10. A method of mixing a sensitive pick-up with insensitive microphones.

Speaker Volume Control.—The individual volume control of speakers presents a problem. A constant impedance fader is unnecessarily expensive, while a variable series resistances for a potentiometer are both unsatisfactory. The scheme of Fig. 11 is a good compromise. If the potentiometer were used alone the tone would be very thin at low volumes, since the speaker impedance is much greater at high than at low frequencies. The condenser partially compensates for this effect by boosting the base. The values indicated are suitable for a 15 ohm speaker and should be altered proportionately for speakers of different impedance. The value of the condenser can be varied to give more or less bass to suit circumstances.

The line between the output transformer and speaker can cause

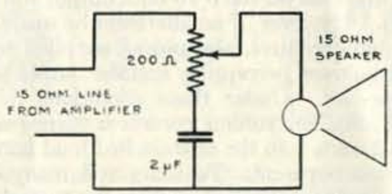
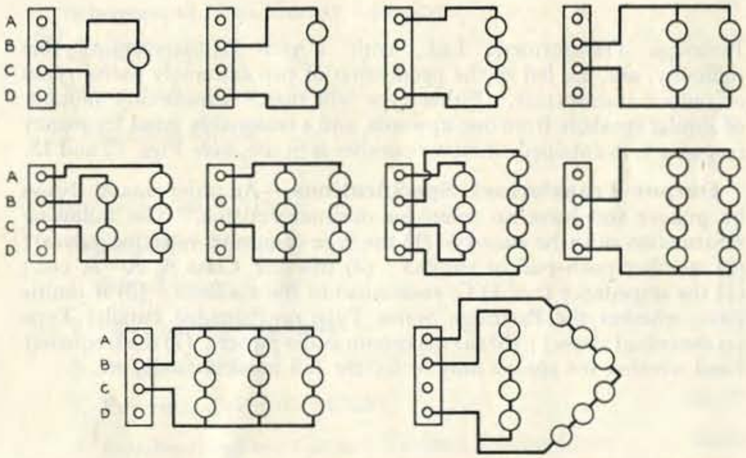


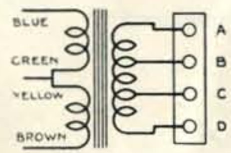
Fig. 11. A tone compensated circuit for controlling the volume of individual speakers.

mis-matching and loss of efficiency unless its resistance is low compared with the impedance of the speaker. Line capacity is not important with the usual low impedance speakers, but its inductance must be kept to a minimum by twisting the two leads (go and return) together, otherwise attenuation of the higher frequencies may occur.

Line Resistance.—Suppose a 2 ohm extension speaker is wired with twin 22 S.W.G. bell wire and is about 10 yards away from the output transformer. Ten yards of twin 22 S.W.G. will have a resistance of around 1 ohm hence the combined impedance of the speaker plus line will be 3 ohms and no less than one-third of the output power will be wasted in the resistance of the line. A line resistance of up to a maximum of 10 per cent. of the speaker impedance is satisfactory.

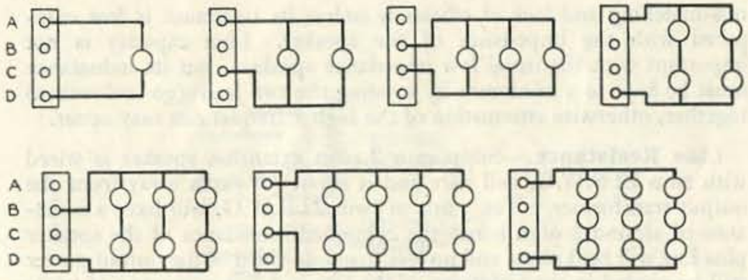
Multi-Ratio Transformers.—A general purpose public address amplifier must be provided with a multi-ratio output transformer so that the full output power can be concentrated upon one or two speakers or else divided amongst a large number of units as the occasion may demand. It is well known that the performance of an output transformer suffers if a wide range of tappings is provided on the secondary. Considerable experimental work has been performed by

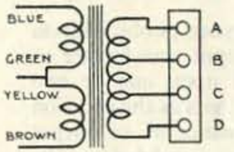




SECTION	OUTPUT IMPEDANCE IN OHMS FOR 1 OHM SPEAKERS	FOR 4 OHM SPEAKERS	FOR 8 OHM SPEAKERS	FOR 15 OHM SPEAKERS	FOR 35 OHM SPEAKERS
AD	3	12	24	45	105
AC	1½	6	12	22½	52½
BD	2	8	16	30	70
BC	1	4	8	15	35

Fig. 12. Method of employing the Series Type Partridge Output Transformer with speakers of equal impedance.





SECTION	OUTPUT IMPEDANCE IN OHMS				
	FOR 1 OHM SPEAKERS	FOR 4 OHM SPEAKERS	FOR 8 OHM SPEAKERS	FOR 15 OHM SPEAKERS	FOR 35 OHM SPEAKERS
AD	1	4	8	15	35
AC	½	2	4	7½	17½
BD	2/3	2⅔	5⅓	10	23⅓
BC	⅓	1⅓	2⅔	5	11⅔

Fig. 13. Method of connecting the Parallel Type Partridge output Transformer, recommended for short lines and when a single speaker is most often used.

Partridge Transformers, Ltd., with a view to overcoming this difficulty, and has led to the production of two extremely useful types of output transformer. Either type will match almost any number of similar speakers from one upwards, and a reasonably good frequency response is maintained whatever number is in use, *vide* Figs. 12 and 13.

Output Transformer Specifications.—An order should always be precise and leave no room for misinterpretation. The following information must be stated :—(1) the type of output valve (or valves) ; (2) whether push-pull or parallel ; (3) whether Class A, A—B, etc. ; (4) the impedance (not D.C. resistance) of the speakers ; (5) if multi-ratio, whether the Partridge Series Type or Partridge Parallel Type (as described above) ; (5) the maximum audio power ; (7) the frequency band whether for speech only or for the full musical range, etc.

SOME USEFUL CIRCUITS, DESIGN TABLES AND CHARTS.

SOME USEFUL CIRCUITS

2w. BATTERY AMPLIFIER

Specification.—Two watts audio output with approximately 0.05v. input. Bass and treble tone control. Entirely battery driven. Will operate direct from a sensitive carbon microphone.

Suggested Use.—For all purposes where mains are not available. Every care has been taken to produce the finest quality possible with a quiescent output stage, and when used with an exponential horn speaker, the same volume of sound will be produced as is normally obtained from a 12 w. mains amplifier working into a baffle mounted speaker.

Technical Notes.—Quiescent battery amplifiers are usually associated with low price and poor quality. For P.A. work a few shillings on the cost is not so important as good quality. The Partridge 2-w. Battery Amplifier is definitely more expensive to build than the average battery job, but the results are so far ahead as to rival mains-driven equipment. The special features are :—(*a*) Liberal decoupling ; (*b*) well designed audio transformers with small magnetic leakage ; (*c*) low distortion damping in grid and anode circuits of the QP240.

Constructional Notes.—The lay-out is not at all critical. T1 and T2 should be placed at right angles to each other, and the grid and anode leads of the QP240 must be kept very short. R10, R11, C7 and C8 should all be connected close to the valve holder of the QP240.

Operational Notes.—The H.T. batteries specified give 180v. when connected in series. Only 150v. is needed at H.T.1, but as the battery voltage falls with use H.T.1 can be plugged into higher and higher tappings in order to keep it at a true 150v. . . . naturally H.T.2 and 3 will require similar adjustments. The method of correctly adjusting H.T.2 and 3 is fully explained in the printed instructions supplied with the QP240.

The L.T. should be obtained from a 2v. accumulator capable of delivering 0·7A continuously.

The grid bias voltages are G.B.1. 1-5v., G.B.2. 3v., G.B.3. 11.5v. These are correct only if H.T.1 is kept at 150v. If the voltage is allowed to fall marked distortion will result.

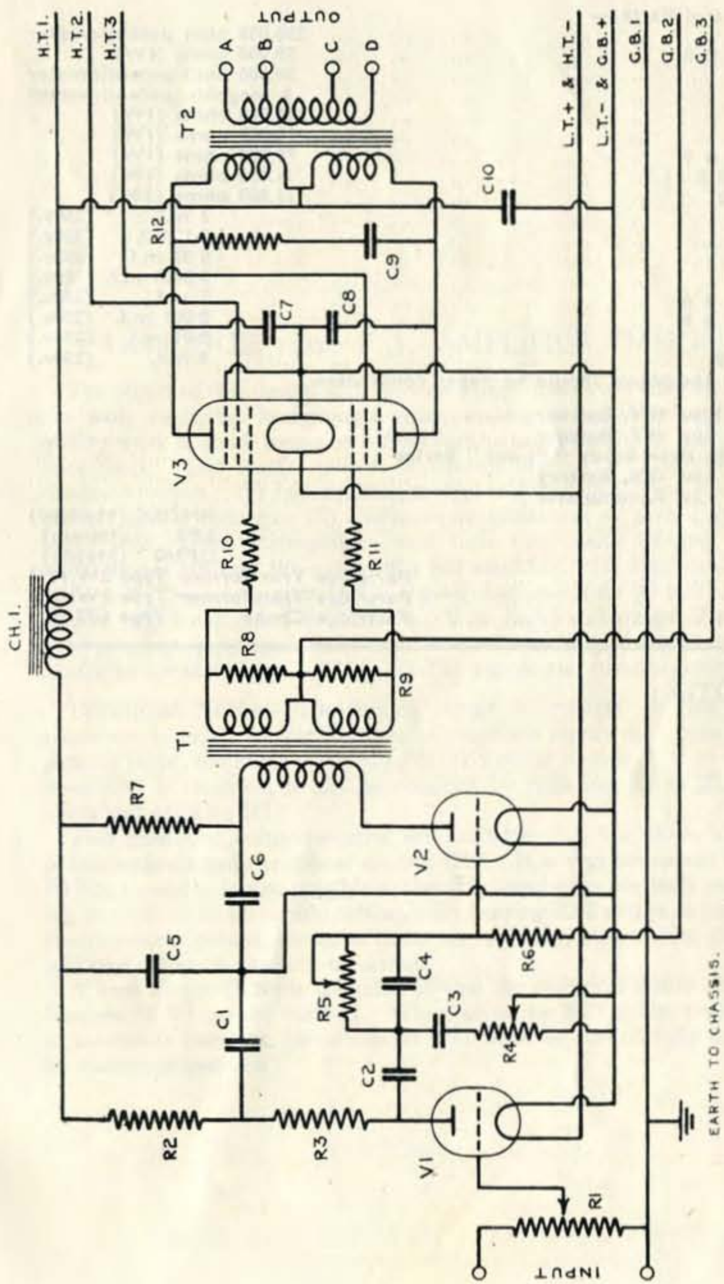


Fig. 14. Partridge 2-watt Amplifier.

List of Parts :—

R1 ...	250,000 ohm potentiometer	
R2 & 3 ...	50,000 ohms (1W)	
R4 ...	50,000 ohm potentiometer	
R5 ...	1 megohm potentiometer	
R6 ...	250,000 ohms (1W)	
R7 ...	10,000 ohms (1W)	
R8 & 9 ...	75,000 ohms (1W)	
R10 & 11 ...	25,000 ohms (1W)	
R12 ...	15,000 ohms (2W)	
C1 ...	2 m.f.	(250v.)
C2 ...	0·1 m.f.	(250v.)
C3 ...	0·02 m.f.	(250v.)
C4 ...	0·0005 m.f.	(250v.)
C5 & 6 ...	2 m.f.	(250v.)
C7 & 8 ...	0·001 m.f.	(250v.)
C9 ...	0·01 m.f.	(250v.)
C10 ...	4 m.f.	(250v.)

All the above should be paper condensers.

1 120v. H.T. Battery ...	
1 60v. H.T. Battery ...	
Both Ever Ready " Power " Series ...	
1 16v. G.B. Battery ...	
1 2v. Accumulator ...	
V1 ...	PM2DX (Mullard)
V2 ...	LP2 (Osram)
V3 ...	QP240 (Mazda)
T1 ...	Partridge Transformer Type 2W/1V/2
T2 ...	Partridge Transformer Type 2W/O/2
Ch1 ...	Partridge Choke Type C75/25

NOTES :

PARTRIDGE 12W. P.A. AMPLIFIER DESIGN

The object of this design is to provide a high quality portable amplifier with negligible background noise, making it suitable for high quality work in small rooms as well as for the usual small hall and outdoor work. The special points of interest are :—(1) 12 watts undistorted output. (2) Independent control and mixing for two microphones and a pick-up. (3) Sufficient amplification to give a good range with carbon microphones, and close range with moving coil microphones without the need for a pre-amplifier. (4) High quality push-pull (Class A) output giving a good response from 30 to 15,000 cycles. (5) Small size (16 in.×9 in.×9 in. high) and weight 27 lbs. (6) Special output transformer enables exact matching of any number of similar speakers from 1—10. (7) The simple and robust quality.

Technical Notes.—The mixing circuit is arranged so that the sensitivity is much greater from the microphone inputs than from the pick-up input, the latter being suitable for a cystal pick-up. If greater sensitivity is required, it can be obtained by reducing R5 to 25,000 ohms and omitting R1.

Two means of volume control are available . . . the three input potentiometers and the master control R12. It is very important that R12 be turned as low as possible so that R2, 3 and 4 are normally working at about their maximum setting. By keeping R12 always as low as circumstances permit, the noise (hum microphomy, etc.) arising in V1 will be kept at an absolute minimum.

A hum-dinger (R20) is provided so that the electrical centre of the filament of V1 can be earthed. While adjusting R20 to the position of minimum hum the potentiometer R12 must be turned fully up to its maximum position.

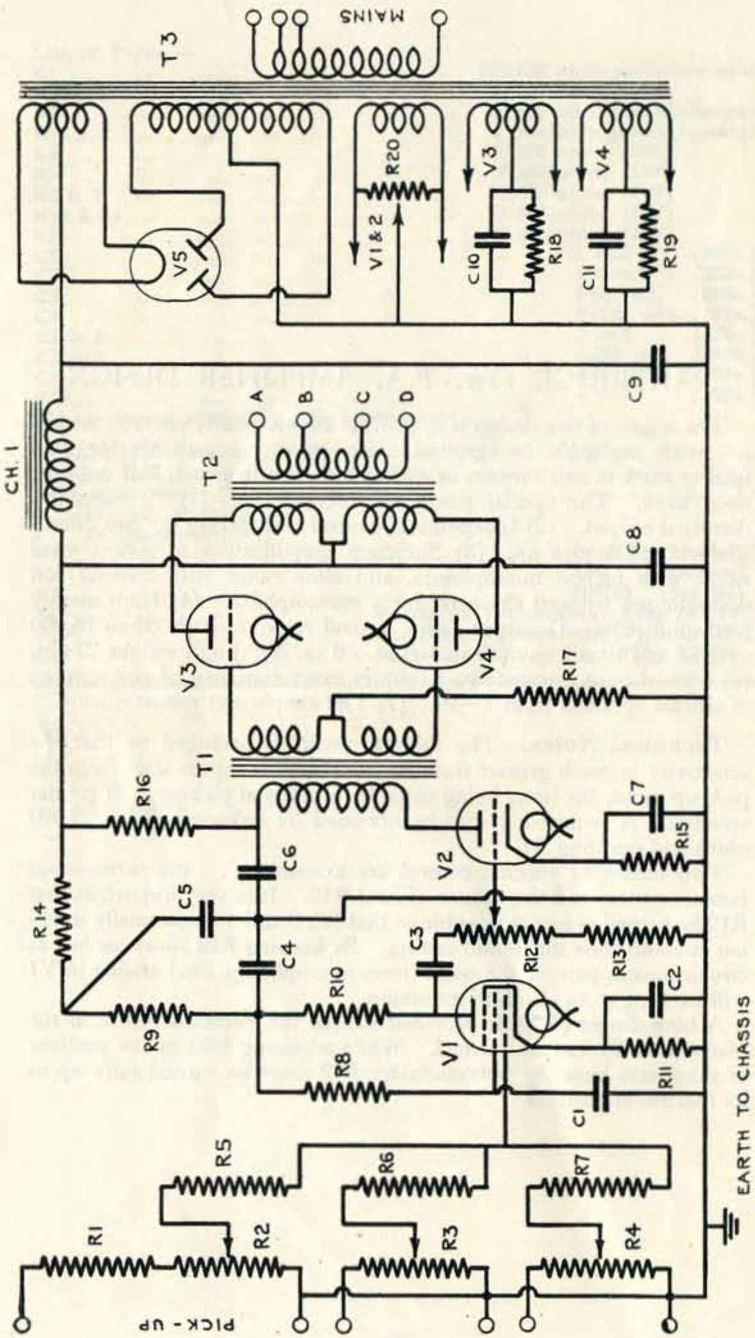


Fig. 15. The Partridge 12 W.P.A. Amplifier.

List of Parts :—

R1	**250,000 ohms (1W)**
R2, 3 & 4	**250,000 ohm potentiometer**
R5	**2 megohms (1W)**
R6, 7 & 8	**250,000 ohms (1W)**
R9 & 14	**40,000 ohms (1W)**
R10	**70,000 ohms (1W)**
R11	**500 ohms (1W)**
R12	**250,000 ohm potentiometer**
R13	**50,000 ohms (1W)**
R15	**600 ohms (1W)**
R16	**35,000 ohms (3W)**
R17	**30,000 ohms (1W)**
R18 & 19	**500 ohms (3W)**
R20	**30 ohm hum-dinger**
C1	**2 m.f. (450v.)**
C2	**50 m.f. (12v. Elec.)**
C3	**0·1 m.f. (450v.)**
C4, 5 & 6	**8, 8 & 8 m.f. (500v.) Dubilier Elec. Block Type 312**
C7	**50 m.f. (12v. Elec.)**
C8	**8 m.f. (500v. Wet Elec.)**
C9	**4 m.f. (650v.) Dubilier Type L.E.G.**
C10 & 11	**50 m.f. (50v. Elec.)**

All condensers not marked Electrolytic (Elec.) must be paper.

V1 ...	**AC/S2 Pen. Mazda (Metallized),**	**2 m.a. H.T. (anode) 0·7 m.a. H.T. (screen)**
V2	**AC/HL Mazda (Plain),**	**6 m.a. H.T.**
V3 & 4	**PP5/400 Mazda (Plain),**	**126 m.a. H.T.**
V5	**5V4 Brimar**	**134 m.a. H.T. (total)**

T1	**Partridge Transformer Type 1V60**
T2	**Partridge Transformer Type 12W/O**
T3	**Partridge Transformer Type 12W/M**
Ch1	**Partridge Choke Type 12W/SC**

NOTES :

PARTRIDGE 45w. AMPLIFIER DESIGN

The Partridge 45-w. circuit has been produced as a result of the many requests from cutomers who wanted an amplifier with great output, mains bias, meters and other refinements that are desirable in equipment of this calibre. It is a useful piece of standard equipment for a P.A. man since several amplifiers can be used in parallel for corresponding increase of power, an arrangement which gives him greater flexibility with the minimum capital outlay.

Specification.—45 w. audio-output with an input of approximately 0·1v. Mains bias from the separate rectifier circuits. Special adjustments for bias voltage and for balancing the output pair to compensate for irregularities in manufacture of the valves. Diode damping of the output grid to prevent excessive positive drive and to protect the power stage from surges.

Spare H.T. and L.T. is provided to feed the Partridge two-stage pre-amplifier, which is particularly suitable for use in conjunction with this Amplifier.

List of Parts :—

R1	**250,000 ohm potentiometer**
R2	**100,000 ohms (1W)**
R3	**50,000 ohms (1W)**
R4	**750 ohms (1W)**
R5	**250,000 ohms (1W)**
R6	**10,000 ohms (10W)**
R7	**550 ohms (1W)**
R8	**50,000 ohms (1W)**
R9	**10,000 ohm potentiometer (wire wound, 10 m.a.)**
R10	**60,000 ohms (1W)**
R11	**70,000 ohms (1W)**
R12	**10,000 ohm potentiometer (wire wound, 10 m.a.)**
R13 & 14	**1,000 ohms (1W)**
C1	**50 m.f. (12v. Elec.)**
C2	**0·1 m.f. (450v.)**
C3 & 5	**2 m.f. (450v.)**
C4	**2 m.f. (650v.)**
C6	**50 m.f. (50v. Elec.)**
C7 & 8	**8 & 8 m.f. (500v.)**
	Dubilier Elec. Block Type 0288 with common positive.
C9	**2 m.f. (450v.)**
C10 & 11	**32 m.f. (320v.) Wet Elec. (T.C.C.)**
A, B & C.	**Sockets for meter jack. These must be designed so that the meter connects before the H.T. supply is broken.**
H.50	**Westinghouse metal rectifier Type H.50.**
Meter (not shown in diagram) 0—250 m.a. moving coil	
V1	**MH41 Osram (Metallized) 2·5 m.a H.T.**
V2	**AC/P Mazda (Plain) 23 m.a H.T.**
V3	**D41 Osram (Plain)**
V4 & 5	**DA30 Osram (Plain) 100 m.a H.T.**
V6 & 7	**5U4 Brimar 125·5 m.a H.T.**
T1	**Partridge Transformer Type 1V240**
T2	**Partridge Transformer Type 45W/O**
T3	**Partridge Transformer Type 45W/M**
Ch1	**Partridge Choke Type 45W/SC**
Ch2	**Partridge Choke Type 45W/FC**
Ch3	**Partridge Choke Type 45W/BC**

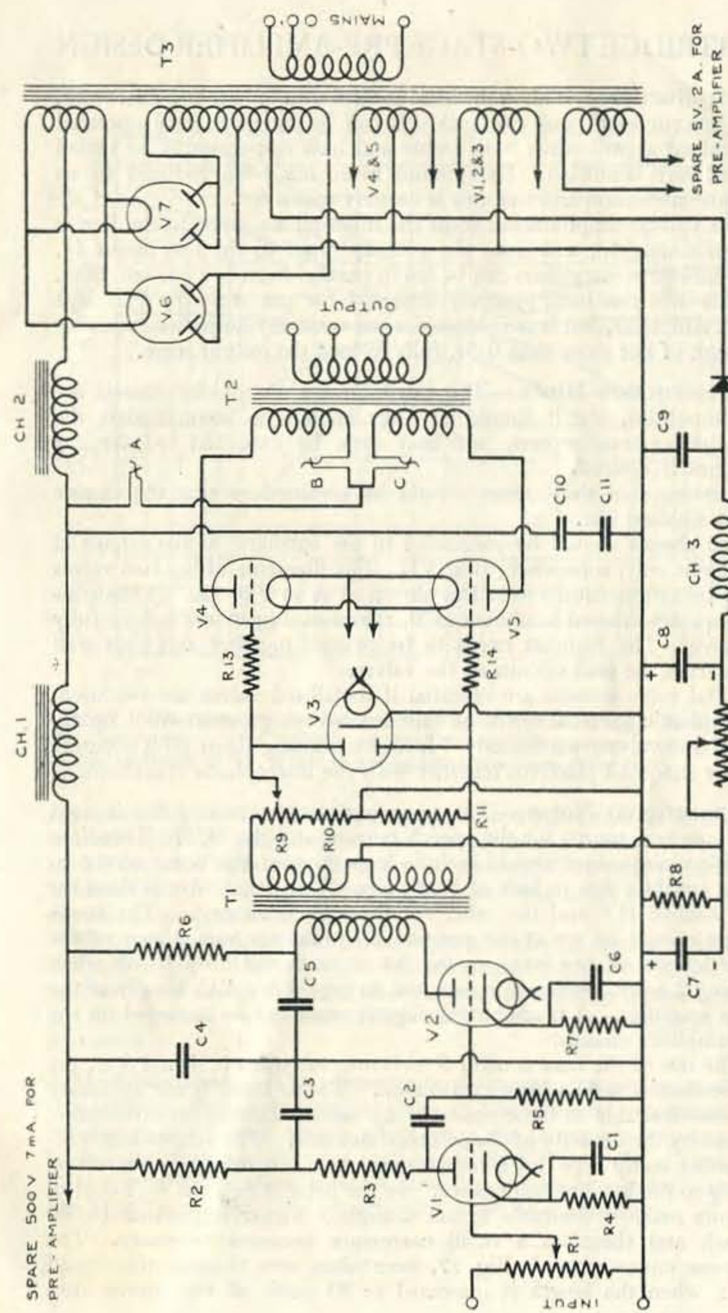


Fig. 16. The Partridge 45w. Amplifier.

PARTRIDGE TWO-STAGE PRE-AMPLIFIER DESIGN

Introduction.—This is the ideal control unit for general P.A. work. Two microphones and two pick-ups can be independently operated and mixed at will while both treble and bass response can be varied to suit any condition. Background noise has been reduced to an absolute minimum and the unit is entirely mains fed.

The voltage amplification from the microphone input to the line is approximately 15, and from the pick-up input to the line about 1½. Any number of amplifiers can be fed in parallel from one pre-amplifier.

This unit has been specially designed for use with the Partridge 45-w. Amplifier, but it is suitable for use with any amplifier requiring an input of not more than 0·5v. fully to load the output stage.

Construction Hints.—The exact dimensions of the chassis are not important, but it should be large enough to accommodate the microphone transformers, and may even be extended to take the batteries if desired.

It is essential the bottom should be screened so that the chassis forms a closed box.

The chassis should be connected to the earthline of the circuit at one point only, somewhere bear V1. The filaments of the two valves must be symmetrically wired as shown at A in Fig. 18. If the connections are crossed as shown as B, the hum dinger will not be fully effective. The filament needs to be twisted together and kept well away from the grid circuits of the valves.

Metal valve screens are essential if metallised valves are not used. Microphonic feed-back will be reduced by tying cotton wool tightly round the valves, particularly V1, and mounting them on a separate rubber mounted platform together with the microphone transformers.

Operational Notes.—The screening of the screened flex is used as a common return for the speech current and the H.T. Therefore routine maintenance should include inspection of this outer screen to avoid troubles due to lack of continuity. One inner wire is used for the positive HT and the other for the speech current. The hum-dinger should be set at the position of minimum hum. Any of the three lengths of flex given in the list of parts will drop 1 volt when passing 2 amps., hence it is essential to give a 5v. A.C. supply at the main amplifier. A filament transformer must not be mounted on the pre-amplifier chassis.

The use of the bass control is obvious, but studs 3, 4 and 5 on the treble control need a little explanation. Treble boost is not necessary or even desirable so these positions are used to correct for attenuation caused by the capacity of the screened flex line. The single-stage pre-amplifier using type L4/1 transformer does not need any correction owing to the low line impedance. In the present case the tone control circuits make it desirable to use a slightly higher impedance (5,000 ohms), and therefore a small correction becomes necessary. The response curves given in Fig. 17, were taken with 15 yards of screened line: when the length is increased to 30 yards all the curves drop

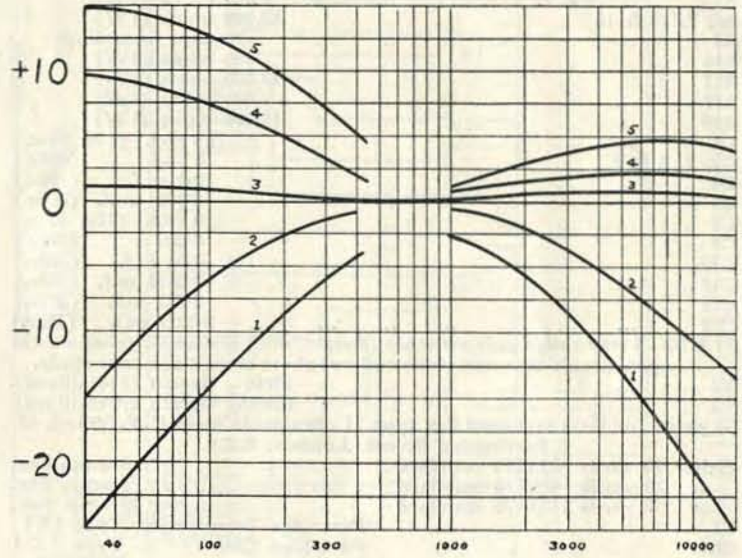


Fig. 17. The frequency response curves of the Partridge Two-Stage Pre-Amplifier. The figures against the treble resonse curves refer to the settings of S1, while those against the bass refer to S2.

approximately 2 db. at 10,000 cycles so that stud 4 gives a level response; similarly with 50 yeards of line the curves drop a further 3 db., and stud 5 becomes the setting for a level response. The total variation is so small (only 5 db.) that an intermediate line length will cause no aural deviation from the level.

R9 controls the gain of the first stage, and should always be kept as low as possible. The two input volume controls should be turned well up (not overloading V1) and the volume adjusted by reducing R9. By this method valve hiss arising in V1 will always be kept at a minimum.

Modifications.—It is strongly urged that no changes be made in this circuit. Relatively small changes in the values of condensers or resistances can have a serious affect upon the frequency response.

Accumulators may be used for the 4V, 2a L.T. supply. This will in fact remove the slight trace of hum that otherwise can be heard in the background when the amplification is set to a maximum.

List of Parts :—

R1, 2, 9, 12 & 15	250,000 ohm potentiometer
R3, 4, 11, 13 & 14	250,000 ohms (1W)
R5	2,000 ohms (1W)
R6, 7, 10 & 18	50,000 ohms (1W)
R8	30 ohm hum-dinger
R16	300 ohms (1W)
R17	20,000 ohms (1W)
R19	5,000 ohms (1W)
R20	15,000 ohms (1W)
C1	50 m.f. (12v. Elec.)
C2, 3, 6 & 7	8 m.f. (450v.)
C4	0·1 m.f. (450v.)
C5	0·0001 m.f. (250v.)
C8	100 m.f. (12v. Elec.)
C9	0·002 (250v.)
C10	0·02 m.f. (250v.)
C11	0·0015 m.f. (250v.)
C12	0·002 m.f. (250v.)
C13	0·0027 m.f. (250v.)

S1 & S2 5 way single pole switches (Bulgin, etc.). These switches should open circuit between studs and not short circuit adjacent studs.

V1	MH4 Osram (Metallized)
V2	MH41 Osram (Metallized)

50 yards (or less) screened flex type 21 obtainable from C. F. Ward, 45, Farringdon Street, London, E.C.4.

Either 10 yards	23/0076 standard	3 amp. flex.
or 20 yards	40/0076 standard	5 amp. flex.
or 50 yards	110/0076 standard	15 amp. flex.
T1	Partridge Transformer	Type L2/1
Ch1	Partridge Choke	Type T.C.1

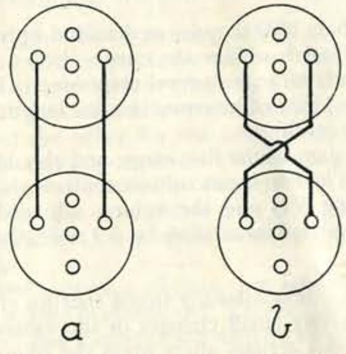


Fig. 18. Two ways of connecting valve filaments. b is not so good as a, where a common hum-dinger is used.

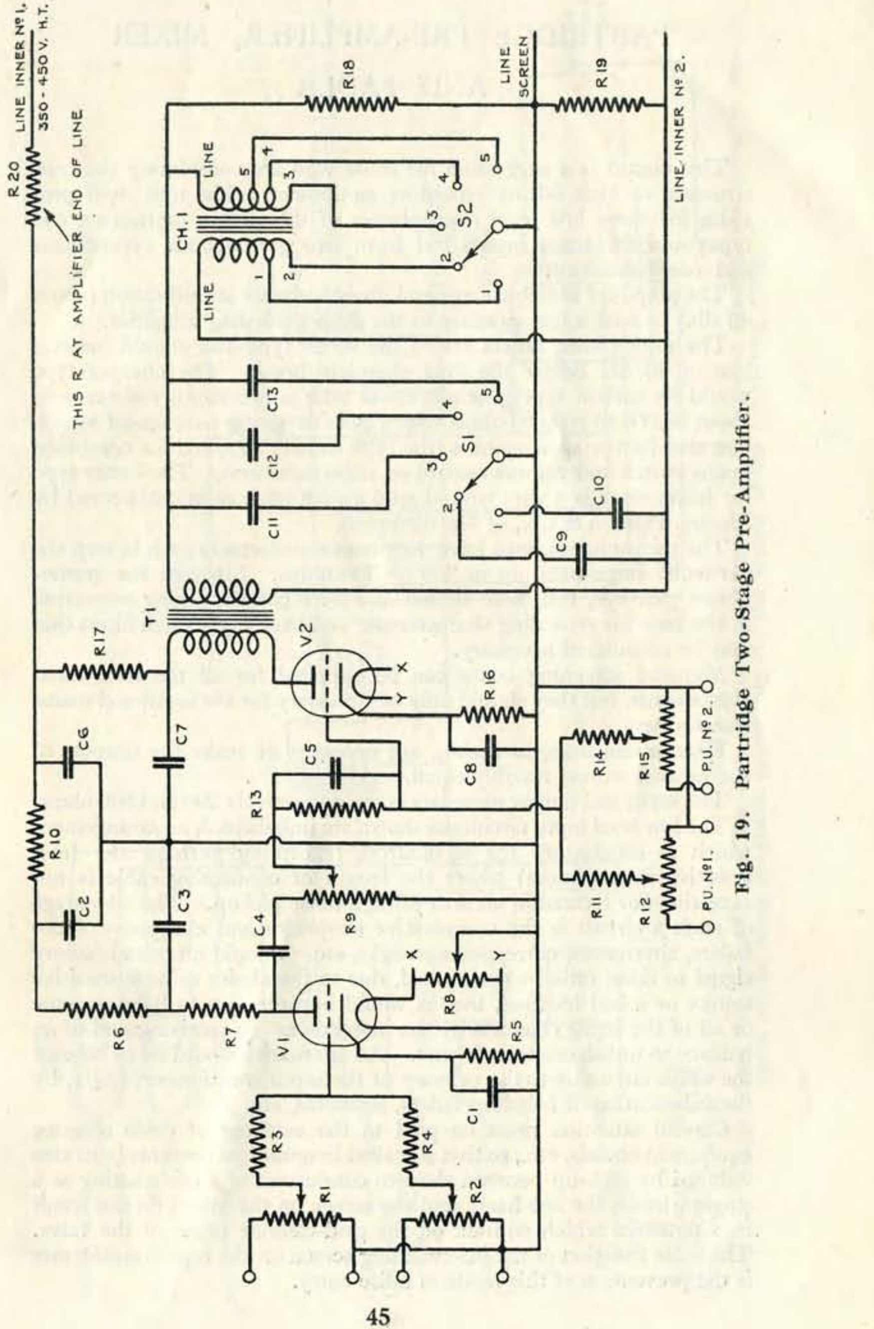


Fig. 19. Partridge Two-Stage Pre-Amplifier.

PARTRIDGE PRE-AMPLIFIER, MIXER AND FADER

This circuit is a suggestion for those who are considering the construction of high fidelity recording equipment. The unit itself provides for three low level microphones of the ribbon or moving coil types and for three inputs fed from two gramophone reproducers and one radio unit.

The amplifier is single stage and gives adequate amplification (about 40 db.) to feed a line running to the main recording amplifier.

The microphone faders are of the series type and should insert a loss of 40 db. before the final complete break. The cheaper type would be carbon type potentiometers with a maximum resistance of about 50,000 to 100,000 ohms with a reliable single pole on-off switch operated by the same spindle (the type usually supplied for combined mains switch and volume control on radio receivers). The better type for heavy duty is a wire wound stud switch such as manufactured by Messrs. Painton & Co., of Northampton.

The microphones must have their own transformers each to step the particular impedance up to 200 or 300 ohms. Likewise the gramophone pick-ups, but these should also have circuits giving correction in the bass for recording characteristic and have any scratch filters that may be considered necessary.

Mumetal screening boxes can be supplied for all the iron cored components, but they should only be necessary for the input and mains transformers.

Fixed attenuators, as shown, are necessary to make the outputs of the various sources roughly equal.

The input and output impedances are respectively 200 and 500 ohms.

The low level input circuits as shown are unbalanced, an arrangement which is satisfactory for all sources (excepting perhaps the least sensitive microphones) where the length of connecting cable is not excessive nor located in an area of high noise pickup. The advantage of such a circuit is the comparative simplicity and cheapness of the faders, alternators, correction networks, etc. Should un unsatisfactory signal to noise ratio be anticipated, due to the choice of an insensitive source or a bad location, then it would be necessary to balance some or all of the input channels by the insertion at the receiving end of an balance to unbalance transformer. An alternative would be to balance the whole circuit up to the primary of the input transformer, LG/1, by the substitution of balanced faders, networks, etc.

Careful attention must be paid to the earthing of cable screens, equipment chassis, etc., so that so-called longitudinal currents (currents induced by pick-up between the two conductors of a cable acting as a single wire on the one hand, and the screen on the other) do not result in a potential which will act on the grid-cathode input of the valve. The main function of the interwinding screen on the input transformer is the prevention of this mode of noise entry.

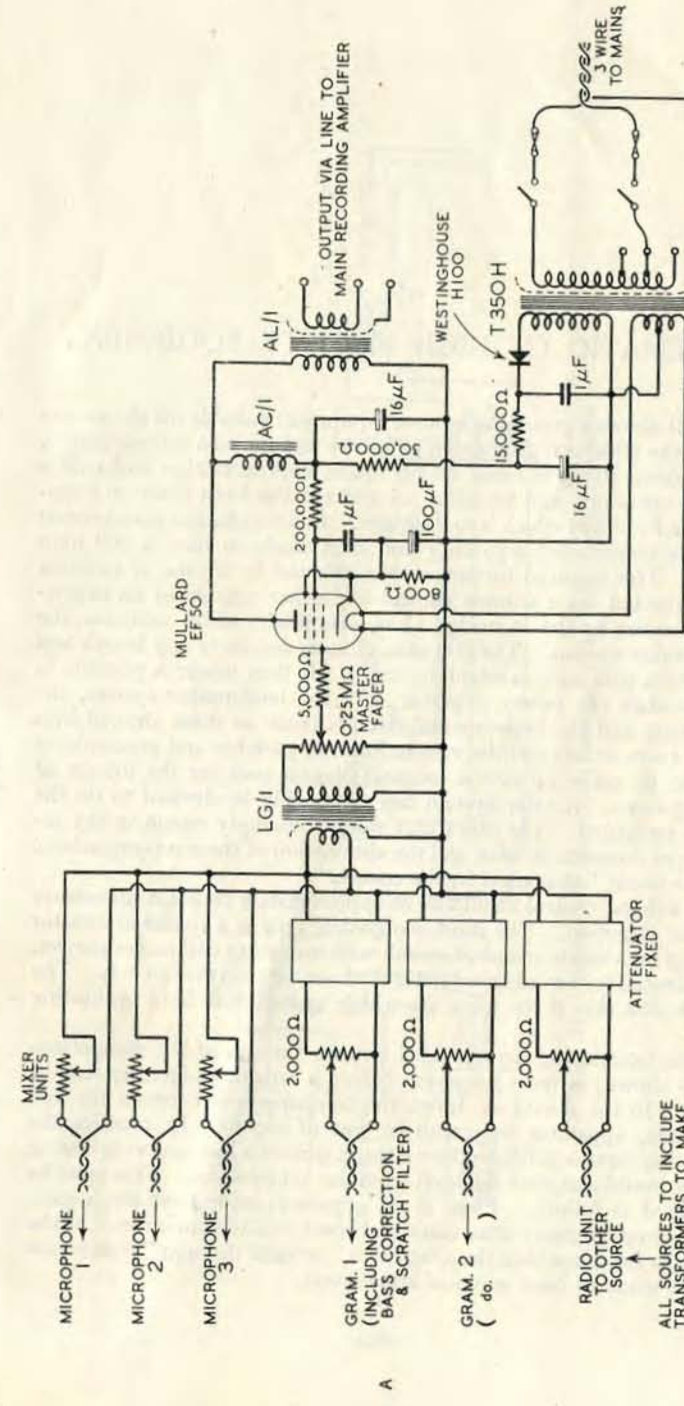


Fig. 20. Single Stage Pre-Amplifier for use with recording equipment. H.T. negative must be connected to the earth of the main amplifier preferably via the screen of the connecting cable.

SCHEMATIC OF HIGH FIDELITY EQUIPMENT

Fig. 21 shows a general scheme of equipment suitable for the person making the faithful reproduction of speech and music a serious hobby. It is realised that the usual set-up makes each alteration and trial a difficult operation, and therefore an attempt has been made to introduce that flexibility which is so desirable. Radio units and gramophone units are constructed separately and each made to have a 500 ohm output. The required output is then selected by means of switches and connected via a volume control and other circuits of an experimental nature to the Partridge 15 w. Amplifier which actuates the loud speaker system. The 500 ohm circuits can be of any length and run in twin wire such as telephone cable. It then becomes possible to accommodate the power amplifier with the loudspeaker system, the radio units and the experimental circuits (such as those shown) in a remote room or loft and the volume control, switches and gramophone unit can be made up into a compact elegant unit for the lounge or drawing-room. A relay system can very easily be devised to do the remote switching. The extra cost would be amply repaid in the reduction of domestic friction and the elimination of those unsympathetic remarks about " that mess in the corner."

The volume control should be of approximately constant impedance in either direction. The most economical type is a ladder attenuator made up on a single arm stud switch with miniature carbon resistances. A balanced type would need naturally a similar twin arm switch. The latter is desirable if the units are widely spaced with long connecting leads.

When introducing experimental circuits the use of the comparison key (as shown) is most necessary before a reliable judgement can be passed. In the circuit as drawn the comparison is between the two conditions, apparatus in circuit or out of circuit. In practice the inserted apparatus will introduce either a gain or a loss and so operating the key would also alter the level from the loudspeaker. This must be eliminated as follows. First, if the apparatus under test has a gain, then a complementary attenuator pad must be added in series. If the apparatus has a loss then the alternative " straight through " connection must be made to have an equal attenuation.

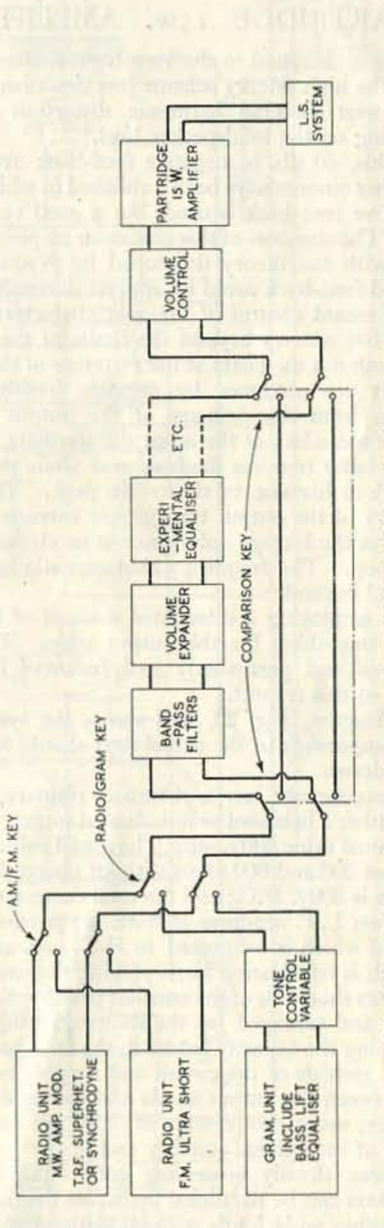


Fig. 21. Schematic for Experimental High Fidelity Equipment.

PARTRIDGE 15w. AMPLIFIER

This amplifier is designed to the very highest standard and to form the nucleus of the high fidelity scheme just described. The aim is to provide the lowest possible harmonic distortion and the highest electrical damping on the loudspeaker load.

To achieve this, 40 db. of negative feed-back over three stages is employed. Other circuits have been published in which a large amount of overall negative feed-back is used, but a good many are not completely stable. The elements of this particular amplifier were calculated in accordance with the theory developed by Nyquist and Bode, so that the required feed-back could be applied to result in unconditional stability. This meant control of the gain characteristic of the feed-back loop over five octaves beyond the limits of the frequency band. In order to smooth out the peaks at the extremes of the frequency band which inevitably are produced by negative feed-back, regenerative voltage is taken from the primary of the output transformer and injected into the secondary of the input transformers, so that the shunt reactance of the latter removes the bass peak while the series reactance of the former does likewise to the treble peak. Taking the voltage from the primary of the output transformer corrects for the iron distortion, but leaves the leakage inductance at an element largely outside the feed-back loop. The frequency characteristic is flat from 30 cps to 15,000 cps and beyond.

Most circuits employing a substantial amount of feed-back require adequate H.T. smoothing for the output stage. This circuit being entirely push-pull and particularly well balanced (self balancing) is much better off in this respect.

The circuit diagram, Fig. 22, well shows the symmetry, and the layout of the components in the actual unit should be made to follow more or less as drawn.

The input transformer has a screened primary, and so can be operated from either a balanced or unbalanced source, the impedance of which has a nominal value 500 ohms. The actual value can, however, be anything between 200 and 600 ohms without observable loss of quality.

The H.T. line is 300v. D.C., and the total current taken of the order of 150 m.a. Two L.T. windings at 6·3v. are needed, the first at 2·4 amps., one leg of which is connected to H.T. neg. and the second at 0.45 amps. which is left floating for supplying the control valve.

All components should be of the smallest possible size consistent with adequate rating and arranged for the shortest wiring. The 1,000 pF condensers forming the capacity ladder in the feed-back path should be mounted on the outside of the circuit and roughly equally spaced one with another between the output anode and the input grid circuits.

Built with care, using components of ± 10 per cent. tolerance, this amplifier is one of exceptional stability and of performance which will delight even those already possessing good equipment. It will be found that speakers can be paralleled up across the output, irrespective of impedance within wide limits without noticeable loss of quality or volume, and that bass resonances in the loudspeakers are effectively damped out.

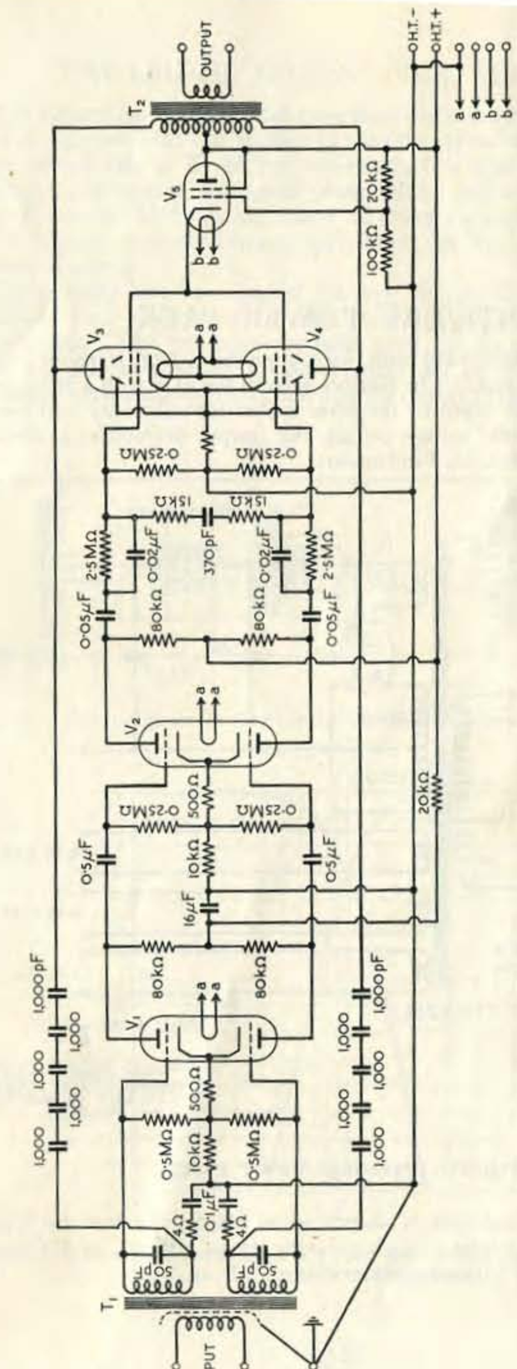


Fig. 22. Partridge 15w. Amplifier, output 15 watts. Gain 65db. Feedback 40 db. over 3 stages. Harmonic Distortion less than 0.1%. Output $Z = 1/30$ output load. Imput $Z = 500$ ohms, or as specified. V_1, V_2, 6SC7, V_3, V_4, 6V6 or 6L6 bias resistor, 120 ohms, or KT66 bias resistor 90 ohms. V_5, TT4 or 6V6.

PARTRIDGE POWER PACK

This is a strightforward unit for supplying a high tension line, 300v. at 150 to 225 m.a. The rectifier valve is the 5U4 with a 5v3 amp. heater. The choke input to the filter makes for reliability and long rectifier life. Ripple voltage across the output terminals is about 0·4 millivolts of 100 c.p.s. fundamental.

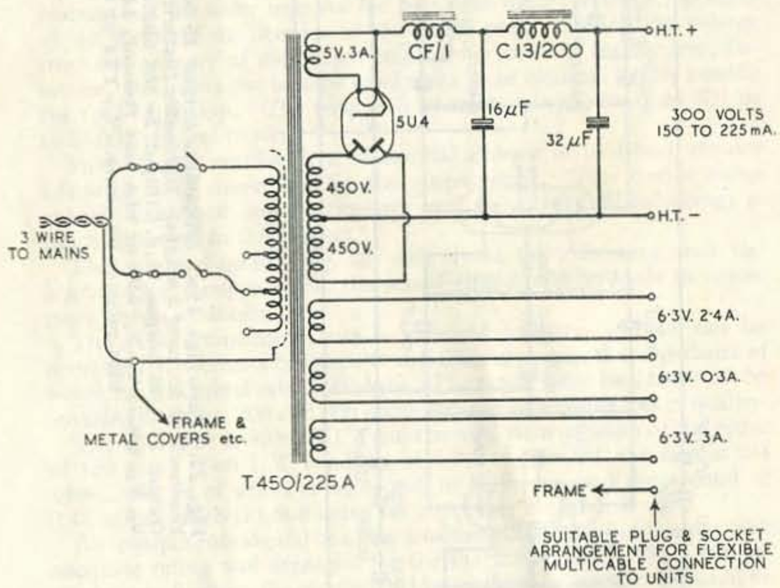


Fig. 23. Partridge Power Pack.

The reservoir condenser is shown as an electrolytic but the type chosen should be robust enough to carry up to 100 ma. of 100 cps ripple continuously without deterioration.

PARTRIDGE CROSS-OVER CIRCUIT

The advantages of this circuit have been discussed earlier in the text. The changeover from one speaker to the other is carried out at a maximum cut-off rate of 12 db. per octave which is a good middle figure not too rapid to give unpleasant phase effects and not too gradual to extend unduly the frequency range of either speaker. Fig. 24 gives the frequency characteristics as calculated and Fig. 26 shows actual measured curves.

Components can be supplied for any values of anode to anode optimum load, output power, cross-over frequency and loudspeaker impedances. The two speakers used need not be of the same impedance. If, however, they differ markedly in sensitivity then a resistance network must be included in the circuit of the more sensitive.

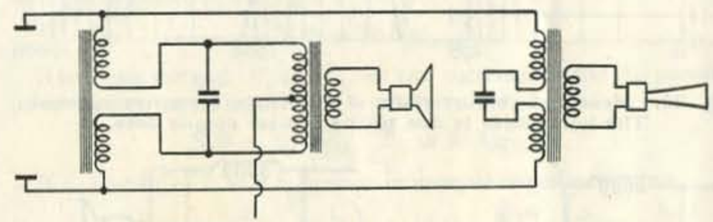


Fig. 24. Partridge crossover circuit developed from that of Fig. 25b.

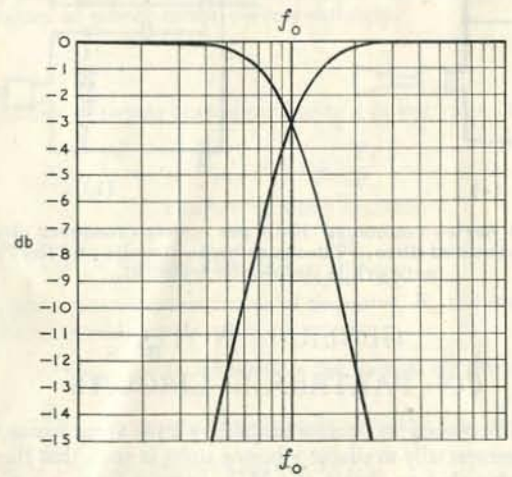


Fig. 25. Calculated curves for a 12 db per octave crossover circuit. ***fo*** **is the crossover frequency.**

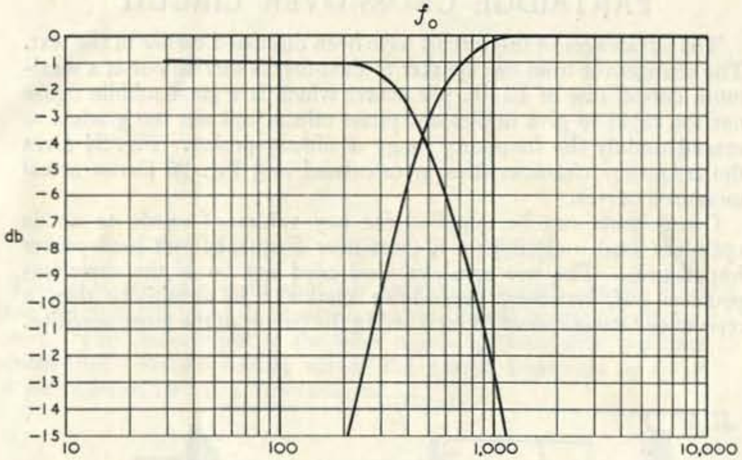


Fig. 26. Measured characteristic of Partridge crossover network. The lower bass is due to the greater copper loss.

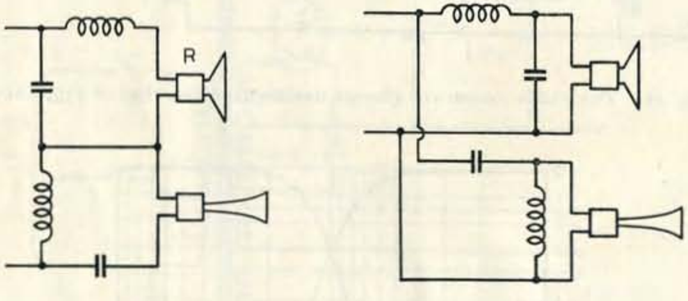


Fig. 27. (a) Series connected 12db per octave crossover circuit and (b) Shunt connected ditto. These are basic circuits and the Partridge network is developed from (b).

GENERAL NOTE ON PARTRIDGE CIRCUITS

It is often necessary to operate amplifiers away from mains, and the design of commercially available vibrator units is such that they can be safely recommended to obtain the H.T. current from a car or other type accumulator. The appropriate transformer must then be ordered with the additional L.T. primary, and the circuit modified to make it possible to supply the valve heaters directly from battery.

DESIGN TABLES AND CHARTS

DESIGN CHARTS

The charts and tables on the following pages are included to complete the usefulness of this manual. These have been selected with due reference to their practical value for the sound engineer and should provide assistance in the rapid computation of circuit elements.

SOME USEFUL FORMULÆ.

Decibel.

The number of decibels N*db* corresponding to the ratio between two amounts of power P_1 and P_2 is

$$Ndb = 10 \log_{10} \frac{P_1}{P_2}$$

When two voltages, E_1 and E_2, or two currents, I_1 and I_2, operate in the same or equal impedances

$$Ndb = 20 \log_{10} \frac{E_1}{E_2}; \text{ or } 20 \log_{10} \frac{I_1}{I_2}$$

If E_1 and E_2 or I_1 and I_2 operate in unequal impedances, then

$$Ndb = 20 \log_{10} \frac{E_1}{E_2} \left(\text{or } \frac{I_1}{I_2}\right) + 10 \log_{10} \frac{Z_2}{Z_1} + 10 \log_{10} \frac{k_2}{k_1}$$

where Z_1 and Z_2 are the magnitudes of the impedances and k_1 and k_2 are the values of power factor correspondingly.

Neper.

The number of nepers corresponding to a power ratio of $\frac{P_1}{P_2}$ is

$$N_{nep} = \tfrac{1}{2} \log_e \frac{P_1}{P_2}$$

$$1 \text{ neper} = 8{\cdot}686 \text{ decibels.}$$

Valve gain.

$$\text{Voltage gain} = \mu \frac{R_L}{R_a + R_L}$$

where μ is the magnification factor of the valve, R_L the load resistance and R_a the valve anode resistance

$$\text{or } G \frac{R_a \ R_L}{R_a + R_L} \text{ where G is the mutual conductance.}$$

When $\frac{R_a}{R_a + R_L}$ is approximately unity as in the case of a pentode or tetrode valve the gain becomes simply $G R_L$.

Valve input capacity.

$$C_{input} = C_{grid\text{-}cathode} + C_{grid\text{-}anode}\left(1 + G\frac{R_a \ R_L}{R_a + R_L}\right)$$

Inductance of iron-cored winding.

$$L = \frac{\mu N^2 \, 4\pi A}{10^9 \, l}$$

where A = mean area of the magnetic path in sq. cms.

l = length of the magnetic path in cms., N = number of turns

μ = the magnetic permiability of the medium.

Optimum gap for Stalloy cores carrying d.c.

Half gap per 1,000 amp. turns of d.c. = 0·03″.

Magnetic Flux density in a transformer core.

$$B = \frac{10^8 \, E}{4{\cdot}44 \, f \, A \, N}$$

where N = number of turns.

A = core area in sq. cms.

f = frequency c.p.s.

E = r.m.s. voltage.

and B = Peak value of A.C. flux density in lines per sq. cm.

Capacity between two parallel plates.

$$C \text{ (picafarads)} = \frac{A}{11{\cdot}3d}$$

where A is the area of the plates, sq. cms. and d is the distance apart, cms.

Capacity between two parallel wires.

$$C \text{ (picafarads per metre)} = \frac{12{\cdot}1}{\log_{10}\frac{d}{r}}$$

where d is the distance between wire centres and r the radius, both in same units.

Reactance of inductance.

$X\!- = 2\pi f L$ ohms where f is the frequency c.p.s.

L the inductance in henrys.

Reactance of a condenser.

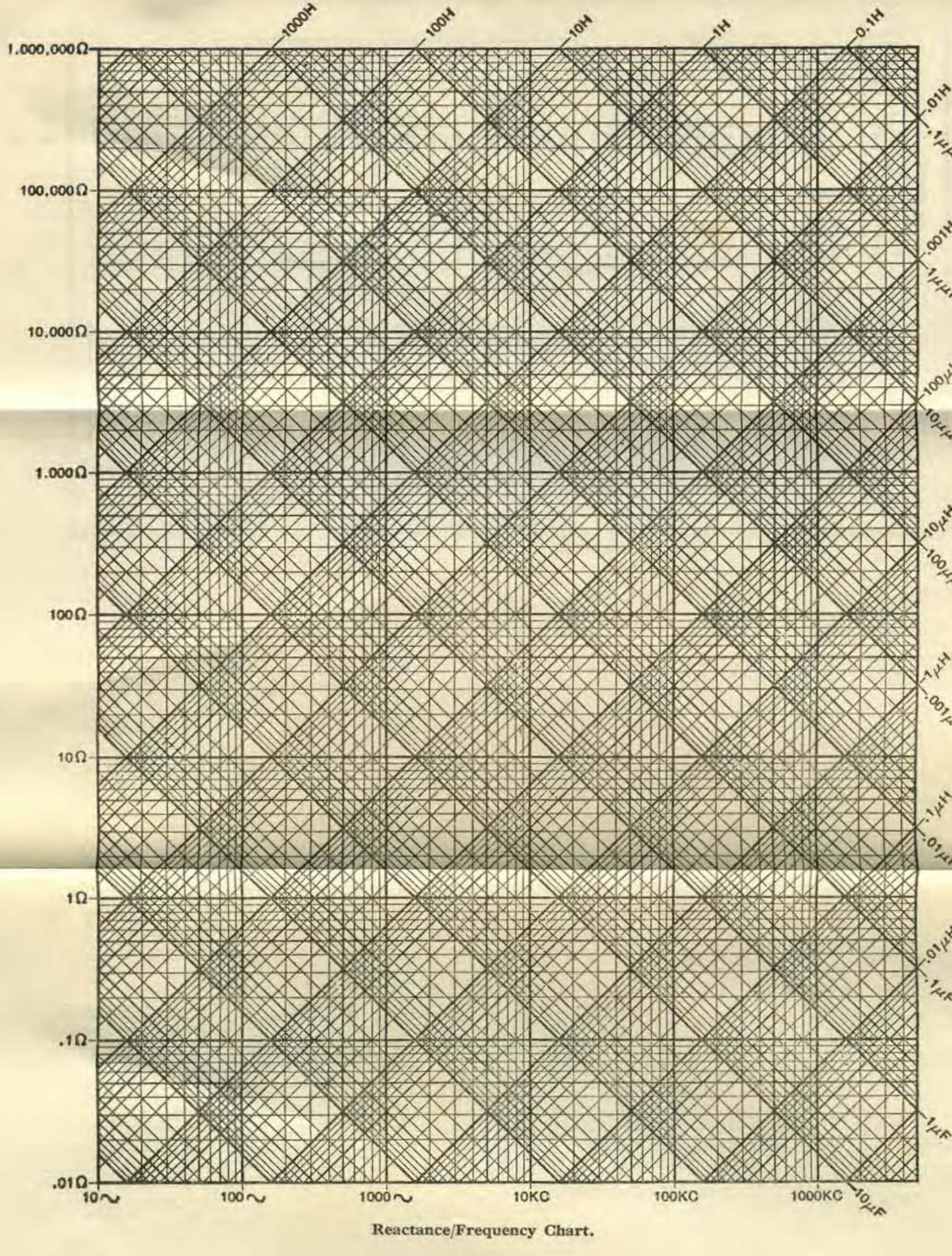
$C\!- = \frac{1}{2\pi f C}$ ohms where C is the capacity in farads.

SOUND POWER TABLE.

Source of Sound.	Maximum Power Radiated. Watts.	Amplifier Output Required to Radiate Equal Power.	
		With Exponential Horn Speakers. Watts.	With Cone Speakers on Large Baffles. Watts.
Normal Conversation	0·0005	0·0015	0·01
Piano	0·15	0·45	3·0
Cinema Organ ..	6·0	18·0	120·0
Small Orchestra ..	5·0	15·0	100·0
Symphony Orchestra	30·0	90·0	600·0
Flute	0·03	0·09	0·6
Trumpet	0·16	0·48	3·2
Cymbals	5·0	15·0	100·0
Brass Drum	12·0	36·0	240·0

DB. CONVERSION TABLE.

Decibels	Voltage or Current Ratio		Power Ratio	
	Up.	Down.	Up.	Down.
1	1·122	0·8913	1·259	0·7943
2	1·259	0·7943	1·585	0·6310
3	1·413	0·7080	1·995	0·5012
4	1·585	0·6310	2·512	0·3981
5	1·778	0·5623	3·162	0·3162
6	1·995	0·5012	3·981	0·2512
7	2·239	0·4467	5·012	0·1995
8	2·512	0·3981	6·310	0·1585
9	2·818	0·3548	7·943	0·1259
10	3·162	0·3162	10·000	0·1000
11	3·548	0·2818	12·59	0·07943
12	3·981	0·2512	15·85	0·06310
13	4·467	0·2239	19·95	0·05012
14	5·012	0·1995	25·12	0·03981
15	5·623	0·1778	31·62	0·03162
16	6·310	0·1585	39·81	0·02512
17	7·080	0·1412	50·12	0·01995
18	7·943	0·1259	63·10	0·01585
19	8·913	0·1122	79·43	0·01259
20	10·000	0·1000	100·00	0·01000
21	11·22	0·0891	125·9	0·00794
22	12·59	0·0794	158·5	0·00631
23	14·13	0·0708	199·5	0·00501
24	15·85	0·0631	251·2	0·00398
25	17·78	0·0562	316·2	0·00316
26	19·95	0·05012	398·1	0·00251
27	22·39	0·04467	501·2	0·00200
28	25·12	0·03981	631·0	0·00159
29	28·18	0·03548	794·3	0·00126
30	31·62	0·03162	1000·0	0·00100
31	35·48	0·02818	—	—
32	39·81	0·02512	—	—
33	44·67	0·02239	—	—
34	50·12	0·01995	—	—
35	56·23	0·01778	—	—
36	63·10	0·01585	—	—
37	70·80	0·01412	—	—
38	79·43	0·01259	—	—
39	89·13	0·01122	—	—
40	100·00	0·01000	—	—
41	112·20	0·00891	—	—
42	125·90	0·00794	—	—
43	141·30	0·00708	—	—
44	158·50	0·00631	—	—
45	177·80	0·00562	—	—
46	199·50	0·00501	—	—
47	223·90	0·00447	—	—
48	251·20	0·00398	—	—
49	281·80	0·00355	—	—
50	316·20	0·00316	—	—



Reactance/Frequency Chart.

TABLE OF POPULAR OUTPUT VALVES.

TYPE.		VOLTS.				CURRENT M.A.				Optimum Load.	Power Output.
		Anode.	Screen.	Grid Bias.		Anode.		Screen.			
Make.	No.			Automatic	Fixed	No. Load.	Full Load.	No. Load.	Full Load.	Ohms.	Watts.
CLASS A (SINGLE VALVE) :—											
Osram	PX25A	400	—	100	—	—	62·5	—	—	4,500	8·4
"	DA30	500	—	134	—	—	60	—	—	4,000	10
"	DA100	1,000	—	146	—	—	100	—	—	6,700	30
"	KT66	250	250	16	—	—	80	—	6	2,200	8
"	KT31	200	180	4·4	—	—	40	—	106	5,500	2·5
Mullard	Pen.428	250	250	12	—	—	72	—	?	3,200	8
"	DO26	400	—	92	—	—	63	—	—	4,000	7
"	EL37	250	250	13·5	—	—	100	—	13·5	2,500	11·5
"	EL38	275	275	9·5	—	—	91	—	11·5	3,000	12·0
Tungsram	P60/500	600	—	110	—	—	130	—	—	2,600	15
American, Tungsram or Brimar	6L6	300	200	12	—	51	54·5	3	4·5	4,500	11
CLASS B WITH GRID CURRENT (PER PAIR OF VALVES) :—											
American, Tungsram or Brimar	46	400	—	—	0	12	120	—	—	5,800	20
" .. Brimar	49	180	—	—	0	4	30	—	—	12,000	3·5
CLASS A—B, NO GRID CURRENT (PER PAIR OF VALVES) :—											
Osram	PX25A	440/400	—	—	117	114	200	—	—	2,800	32
"	DA30	500/480	—	—	145	100	220	—	—	3,400	45
"	DA100	1,000	—	146	—	200	212	—	—	8,000	90
"	KT66	250	250	18	—	120	150	7	17	4,000	17
"	KT66	400	300	—	27	100	165	4	19	5,700	38
"	KT31	200	180	6	—	45	57	11	19	8,000	6
Mullard	Pen.428	375	275	17·5	—	96	124	10	18	6,500	28
"	DO26	440/400	—	—	115	110	200	—	—	2,800	32
"	EL37	325	325	25·0	—	154	180	19·5	60	4,000	35
Tungsram	P60/500	900	—	—	250	60	180	—	—	12,000	70
American, Tungsram or Brimar	6L6	400	300	23·5	—	112	128	7	16	6,600	32
CLASS A—B, WITH GRID CURRENT (PER PAIR OF VALVES) :—											
Osram	DA100	1100/1000	—	—	150	200	330	—	—	8,000	200
American, Tungsram or Brimar	6L6	400	300	—	25	102	230	6	20	3,800	60

EQUALISER OR TONE-CONTROL DESIGN CHART

Equalisers or tone-control circuits using a single reactance (a condenser or inductance) in combination with resistances can be quickly and easily computed by the use of this chart. Such circuits can first be reduced to the fundamental equivalents shown in (1) and (2), X being the reactance of either a condenser or inductance, R_s the effective driving or source resistance and R_l the effective load resistance. Each curve of the family of curves represents the frequency characteristic of the circuit for a particular value of r, which value determines the maximum attenuation at a frequency such that X is either too large or too small to be effective.

The frequency scale is read off in terms of K so that for inductances the frequency increases from left to right for (1) and from right to left for (2). Similarly for condensers the scale reads from right to left for (1) and from left to right for (2).

The procedure is :—

(1) To determine the maximum attenuation to be introduced or in other words the curve on the chart which is desired. This gives the value of α from which r can be calculated.

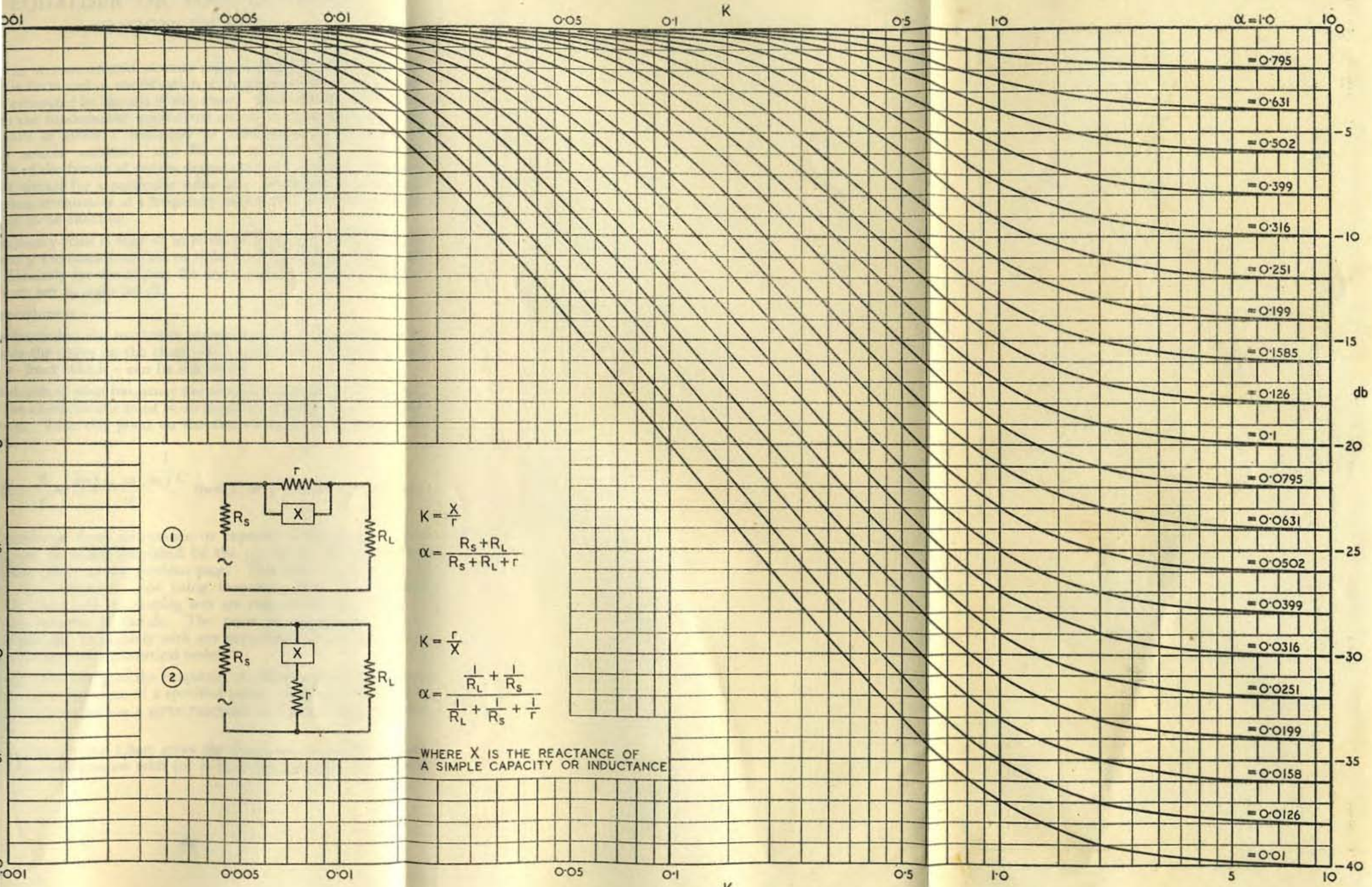
(2) To decide at what frequency the change of attenuation is desired, *e.g.*, that the characteristic must be so many db. down at such and such a frequency. Take this point on the desired curve and read off the value of K.

$$\text{Since } K = \frac{X}{r} = \frac{2\pi f L \text{ or } \dfrac{1}{2\pi f C}}{r} \text{ then L or C can be worked out.}$$

The conversion from inductance or capacity to reactance at any frequency can be much simplified by the use of the Reactance/Frequency Chart given on the previous page. This chart has four sets of ordinates, the horizontal scale being frequency, the vertical being reactance in ohms and the sloping sets are respectively inductance in Henrys and capacity in m.f.ds. The point of intersection of any value of inductance or capacity with any particular value of frequency, will read reactance on the vertical scale.

The Chart will also give the frequency at which a given inductance or capacity has a reactance of a specified value. Similarly the inductance or capacity which has a given reactance at a given frequency can be read off.

A further use of the Chart gives the frequency at which a given inductance is in resonance with (or is of equal reactance to) a given capacity.

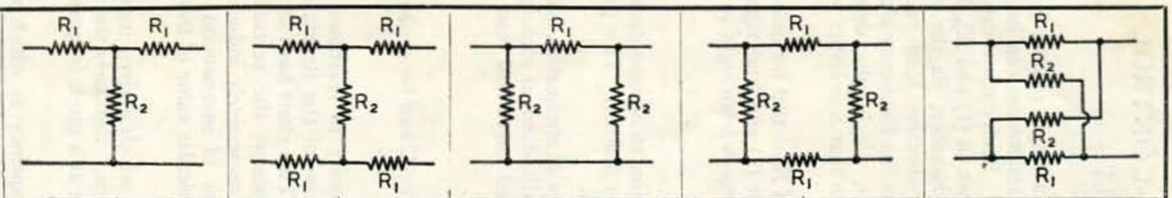


Equaliser Design Chart.

ATTENUATOR CHART.

This gives the elements of various forms of 600 ohm pad for different values of attenuation. For other impedances alter values proportionately, *i.e.*, elements of 300 ohm pad will be ½ those for 600 ohms, 1,200 ohms double, and so on.

NETWORK →



Loss db.	R_1 ohms.	R_2 ohms.	R_1 ohms.	R_2 ohms.	R_1 ohms.	R_2 ohms.	R_1 ohms.	R_2 ohms.	R_1 ohms.	R_2 ohms.
0	0	—	0	—	0	—	0	—	0	—
1·0	34·5	5208	17·25	5208	69·1	10440	34·5	10440	34·5	10440
2·0	68·8	2582	34·4	2582	139·4	5232	69·7	5232	68·8	5232
3·0	102·7	1703	51·3	1703	211·3	3505	105·6	3505	102·7	3505
4·0	135·8	1249	67·9	1249	285·7	2651	142·9	2651	135·8	2651
5·0	168·1	987·6	84·1	987·6	364·5	2141	182·3	2141	168·1	2141
6·0	199·3	803·4	99·7	803·4	447·9	1807	224·0	1807	199·3	1807
7·0	229·7	685·2	114·8	685·2	525·3	1569	262·7	1569	229·7	1569
8·0	258·4	567·6	129·2	567·6	634·2	1393	317·1	1393	358·4	1393
9·0	285·8	487·2	142·9	487·2	738·9	1260	369·4	1260	285·8	1260
10·0	312·0	421·6	156·0	421·6	854·1	1154	427·0	1154	312	1154
12·0	359·1	321·7	179·5	321·7	1119	1002	559·5	1002	359·1	1002
14·0	400·4	249·4	200·2	249·4	1443	899·1	721·5	899·1	400·4	988·1
16·0	435·8	195·1	217·9	195·1	1847	826	923·2	826·0	435·8	826·0
18·0	464·0	152·5	232·0	152·5	2344	772·8	1172	772·8	464·0	772·8
20·0	490·4	121·2	245·2	121·2	2970	733·3	1485	733·3	490·4	733·3
22·0	511·7	95·9	255·9	95·9	3753	703·6	1877	703·6	511·7	703·6
24·0	528·8	76·0	264·4	76·0	4737	680·8	2369	680·8	528·8	680·8
26·0	542·7	60·3	271·4	60·3	5970	663·4	2985	663·4	542·7	663·4
28·0	554·1	47·8	277·0	47·8	7524	649·7	3762	649·7	554·1	649·7
30·0	563·2	37·99	281·6	37·99	9309	639·2	4655	639·2	563·2	639·2
32·0	570·6	30·16	285·3	30·16	11930	630·9	5967	630·9	570·6	630·9
34·0	576·5	23·95	288·3	23·95	15000	624·4	7500	624·4	576·5	624·4
36·0	581·1	18·98	290·6	18·98	18960	619·3	9480	619·3	581·1	619·3
38·0	585·1	15·11	292·5	15·11	23820	615·3	11910	615·3	585·1	615·3
40·0	588·1	12·00	294·1	12·00	30000	612·1	15000	612·1	588·1	612·1