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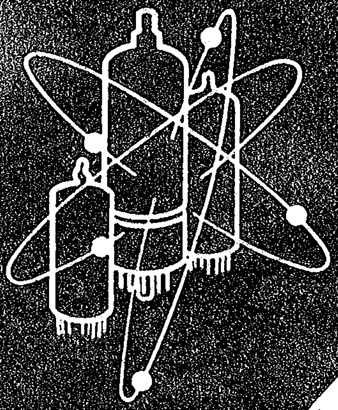

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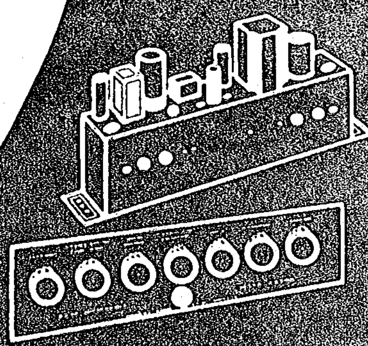

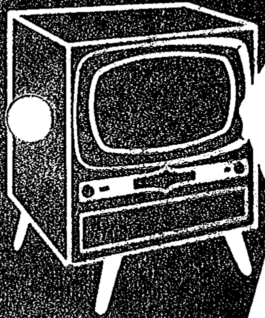

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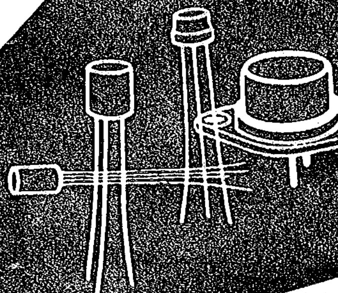

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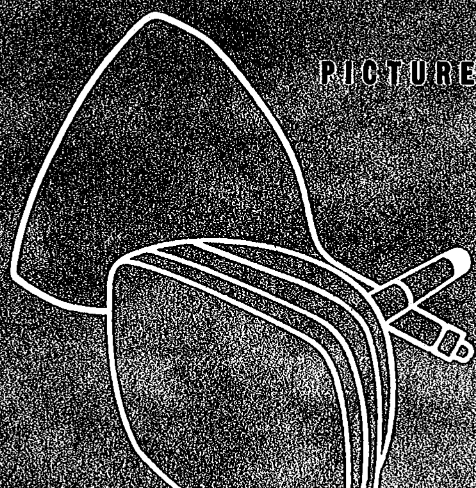

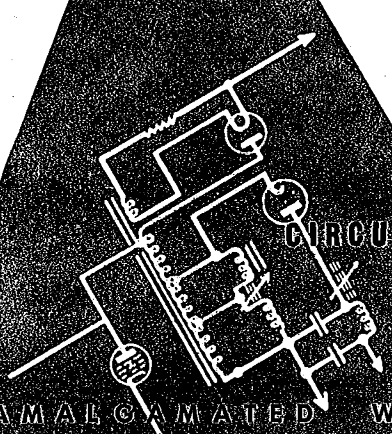

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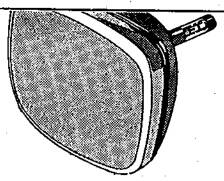

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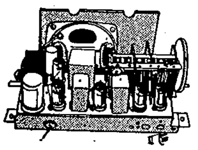

HIGH FIDELITY — PART 1 — GENERAL CONSIDERATIONS 71

This is the first of a 5 part article which discusses the elements of "Hi-Fi" recording and reproduction. It is intended for both the music lover and the technician, and will prove an invaluable guide for newcomers to "Hi-Fi".

VIBRATORS 76

A description of their operation and application.

A RATIO DETECTOR FOR F-M BROADCAST RECEIVERS 79

This article discusses the use of GEX34 germanium diodes in a ratio detector for the reception of F-M broadcasts. Experimental F-M broadcasting stations are operated by the ABC in four capital cities.

EDITOR .... **BERNARD J. SIMPSON**

Radiotronics is published twelve times a year by The Wireless Press for Amalgamated Wireless Valve Company Pty. Ltd. The annual subscription rate in Australasia is 10/-; in U.S.A. and dollar countries $1.50, and in all other countries 12/6. Price of a single copy is 1/-.

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HIGH FIDELITY

PART 1

GENERAL CONSIDERATIONS

Sound is noise. There are some very beautiful noises, as well as some that are very raucous. To some people, even beautiful music sounds only like noise. From the beginning of the talking machine era, when the reproduction of sound became possible, the engineer has been trying to reproduce actual sounds exactly as they were originally produced. This has not been possible until recently for many reasons. Almost any reproducing instrument injects distortions of a number of types into its reproduction of sound. In discussing high fidelity, sound reproduction is being referred to in which these distortions are kept below levels at which they would be audible to a large percentage of the listening public.

One of the first considerations, in thinking of sound reproduction, is what is to be most desired in this reproduction. Should the actual sound be exactly reproduced? Or, should an attempt be made to improve upon that actual sound? This is a subject which must be decided by the listeners themselves. For many years, demonstrations and tests have been made for large groups of listeners in order to arrive at an answer to this problem.

The early tests seemed to indicate that the average listener preferred reproduction with a frequency range which was restricted rather than the type of reproduction which has come to be known as "High Fidelity". However, this has definitely been proven to be a fallacy. One of the major reasons was the fact that the sound modulation in A-M radio broadcasting is generally restricted to a high frequency limit of 5000 c/s, and the average listener became used to listening to responses limited to this. Also, many home instruments were limited in bass response, and in order to make his instrument sound as bassy as possible, the average customer adjusted the high frequency tone control to limit high frequency response still more. Records during this period were made of a material which distinctly limited high-frequency response because surface noise covered up response above 4000 or 5000 c/s.

Until recently, magnetic recorders capable of recording wide-frequency response were not available.

Now, with improved techniques, equipment and improved record material and magnetic tape, the listening desires of the average person have been decidedly changed. They are demanding much more accurate reproduction of sound and because of this, there has been an appreciable increase of interest in high fidelity. In addition to better techniques and equipment, the education of the average person's listening habits have also been a large factor in this demand for high fidelity. Now, it is becoming so that the public expects its reproducing equipment to be capable of exactly duplicating the actual sound originally made, or even to improve upon it.

THE NATURE OF SOUND

Before going into the subject of high fidelity, it may be well to consider the nature of sound.

The earth is surrounded by a deep ocean of air, and we mortals crawl around on the surface of the earth's crust at the bottom of this ocean of air. The air above the earth has a certain weight which bears down upon us and makes it necessary for us to live under constant pressure. At sea level, this pressure is approximately fifteen pounds per square inch. At higher altitudes, such as on mountain peaks, the pressure is less.

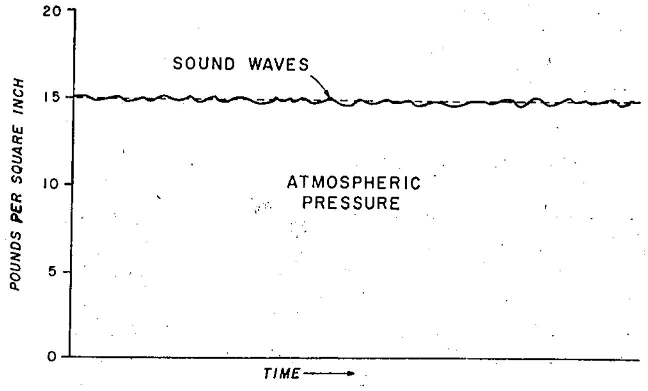


Fig. 1. The nature of sound.

Figure 1 shows a chart of air pressure in which the median line represents fifteen (15) pounds per square inch. Superimposed upon this line are an infinite number of minute variations. The variations have been exaggerated for the purpose of explanation but in reality, they are very minute. If drawn to scale, they would be scarcely noticeable. The variations shown here are sound waves and are caused by the movement of the molecules of air. An example of the movement of molecules of air is when the hands are clapped. Air is violently forced from between the hands, and the air having been forced out, causes a compression wave of air molecules to travel outward from the point of disturbance. This pressure wave travels outwardly at a constant velocity, and can be heard as sound at a distance point if an ear is present.

Sound is a sensation, produced through the ear when certain vibrations are set up in the surrounding air by a vibrating body. In other words, the wave in the air is strictly a wave, and if there is no ear for this wave to impinge upon, no sound will exist. Sound waves have two important characteristics; frequency, which determines the pitch of the sound; and amplitude, which determines the intensity of the sound.

PROPAGATION OF SOUND WAVES

In the upper left corner of figure 2 is seen a loudspeaker giving forth a sound of constant frequency. The cone in the loudspeaker vibrates back and forth pushing the air back and forth with it.

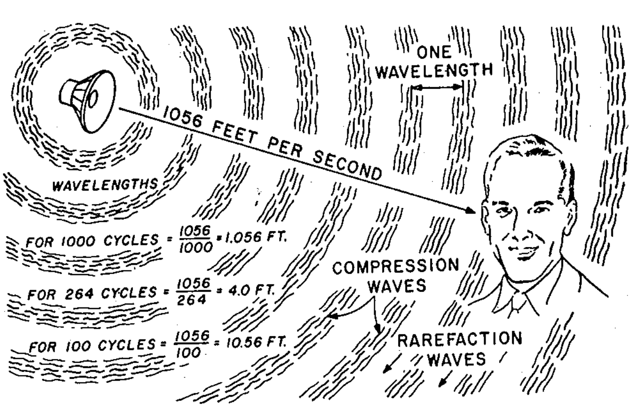


Fig. 2. Propagation of Sound Waves.

When the cone moves outward it presses the air in front of it, and when the cone moves backwards, it causes rarefaction, or a partial vacuum, in the air. These pressure waves and rarefaction waves travel outwardly as shown. The lines indicate the pressure waves and the spaces between the lines indicate the vacuum or rarefaction waves. The listener at the right of the illustration senses these pressure waves when they strike the membranes in his ear. These variations in pressure are interpreted by his ear as sound. These waves travel outwardly from the source of sound at a constant velocity which is approximately 1,056 feet per second. This figure varies somewhat with atmospheric conditions such as temperature and barometric pressure. Because the rate of travel is constant, it will be seen that the distance between waves will be less for a high-frequency tone, the cone of the loudspeaker would vibrate rapidly, and one wave would not travel far from the loudspeaker before the next wave started. The distance between the sound waves is known as the wavelength. One wavelength has been depicted in the upper right corner of the diagram. Knowing the rate of travel of sound and its frequency, the wavelength can be readily computed. For 1000 c/s it would be 1,056 divided by 1,000 which gives slightly over one foot. The wavelength for middle "C" on the piano, which is 264 c/s, is four feet. For 100 c/s the wavelength is 10.5 feet. It can be seen from this that the wavelengths of sound are considerably shorter than radio wavelengths.

Although the vibrations of sound are much slower travelling than radio vibrations, they have a shorter wavelength due to their lower velocity. Sound waves travel only 1,056 feet per second, while radio waves travel 186,330 miles per second.

THE HUMAN EAR

It may be instructive, as well as interesting, to consider briefly the mechanical operations which take place inside the human ear. Figure 3 shows a cross section of the ear. On the left side of the illustration is shown the outer portion of the ear. This is known as the Pinna. When you were a small boy it was your Pinna which your mother insisted that you wash behind. From the centre of the Pinna there is a tube, going into the head, which carries the sound into the ear. The Pinna, however, is not a very efficient sound collector, and a definite improvement can be noticed when the cupped hand is placed behind the ear. This assists considerably in intensifying the sounds heard. The tube leading into the head is more familiar because it is this tube which generally collects wax. Across the end of this tube is a membrane known as the "ear drum". Variations of air pressure cause the ear drum to vibrate, and these vibrations are passed on by suitable mechanisms to the nerve centres. In order that changes of barometric pressure will not affect the ear drum, a tube from the inner side of the ear drum leads directly to the mouth. This tube

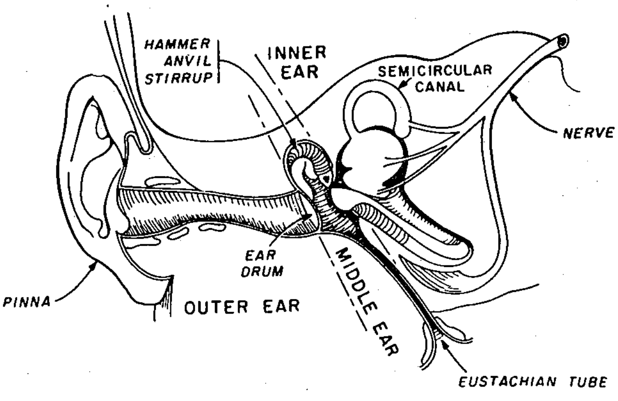


Fig. 3. The Human Ear.

is called the "eustachian tube." Sound vibrations do not travel up the eustachian tube conveniently because it is very small and does not have direct access to the outer air.

Sometimes during illness this tube becomes closed, causing an impairment of hearing. It has no doubt been noticed that when a sudden change of elevation is made, such as going up or down in an elevator, there is a peculiar sensation in the ear. This can be usually relieved by opening the mouth widely, or swallowing. These operations have a tendency to open the eustachian tube and allow an equalization of air pressure on both sides of the ear drum.

Attached to the ear drum is a series of small bones known as the "hammer", "anvil", and "stirrup". These small bones are vibrated by the ear drum and pass the vibrations along to another membrane which encloses a cavity filled with liquid. It is within this liquid that the nerve ends pick up the sensation of sound. These nerve endings are small hairlike devices which protrude into the liquid from the sides of the spiral canal shown in the illustration. Above this, and to the left, will be seen what is known as the semicircular canal of the ear. The semicircular canal has no connection with the function of hearing, but does influence the sensation of stability and equilibrium of the body.

It will be noticed that the ear can be divided into three definite sections. These are known as the outer ear, middle ear and the inner ear. The outer ear is the section from the ear drum out. It contains air, which is subject to sound vibrations. The middle ear is not subject to sound vibrations because the air it contains arrives there in a round-about course and changes pressure only slowly. The sound vibrations are carried across this middle ear by the means of mechanical vibrations in the small bones, the "hammer", "anvil", and "stirrup".

The inner ear is shown to the right of the middle ear and is closed off by membranes. It is filled with a liquid which transmits the sound from the membranes to the nerve endings. The large bundle of nerves which carry the sound sensations to the brain can be seen leading off at the upper right of the illustration. This nerve, when dissected and seen under a microscope has a very striking resemblance to a telephone cable which has a large number of wires bundled together. Just how these impulses of sound or sound sensations are carried to the brain and registered as sound is largely a guess. This general picture of how the ear works will aid in interpreting the various facts which will be discussed later.

RELATIVE RESPONSE OF THE EAR

The intensity of the sound waves determines the amplitude of the vibration in the liquid of the inner ear, and therefore the degree of response of the nerves leading from the liquid. The response to sounds of different intensities is proportional to the logarithm of the intensity rather than directly to the intensity. Thus, if there are three sounds with relative intensities of 1, 10 and 100 units, the ear will perceive the same relative difference in intensity between the one and ten unit sounds as between the sounds having relative levels of ten and one hundred units. The ear is interested only in the ratio between sounds, not the actual level. In the case just related, the 1 and 10 unit sounds and the 10 and 100 unit sounds had a ratio of ten to one in each case.

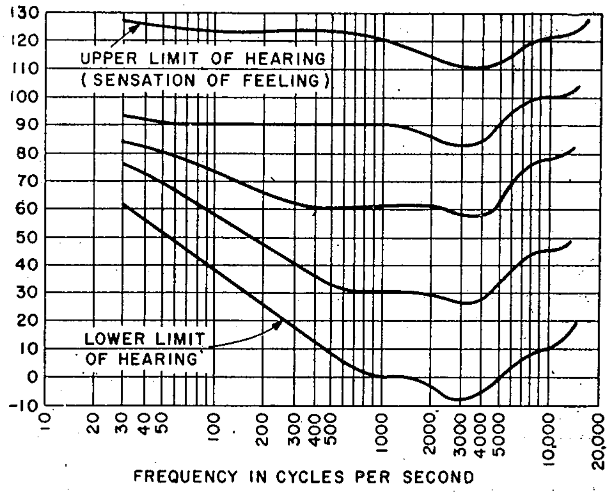
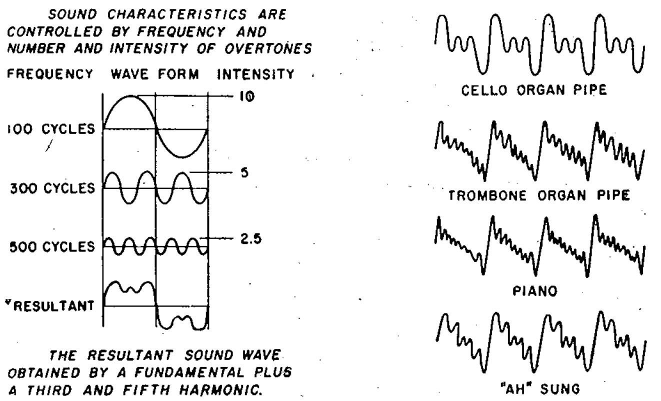


Fig. 4. Response of the Ear.

The response of the ear varies both with the frequency and intensity of the sound. Figure 4 shows the response of an average ear at different frequencies and intensity levels. These curves have been experimentally determined and show that the response of the ear is reasonably flat at high intensity levels but droops appreciably at certain frequencies at lower intensity levels. This is the reason that compensated volume controls, or loudness controls are used in almost all instruments (high-fidelity and otherwise) today. These curves are based on equal loudness. Each curve is plotted through the point (db) for each frequency where the level of sound sounds the same.

QUALITY OF SOUND

A chart of the variations of air pressure caused by a single frequency note will consist of a smooth curve known as a sine wave. Such a sine wave is shown at the top left of figure 5. However, such a note is not commonly found in mechanically produced sounds normally found in music or speech. Most sounds contain not only a basic frequency, but also harmonics of that frequency. For instance, on the left of this chart, at the bottom, a type of sound wave is shown which more nearly conforms to the usual type of sound waves produced by musical instruments.



SOUND CHARACTERISTICS ARE CONTROLLED BY FREQUENCY AND NUMBER AND INTENSITY OF OVERTONES

THE RESULTANT SOUND WAVE OBTAINED BY A FUNDAMENTAL PLUS A THIRD AND FIFTH HARMONIC.

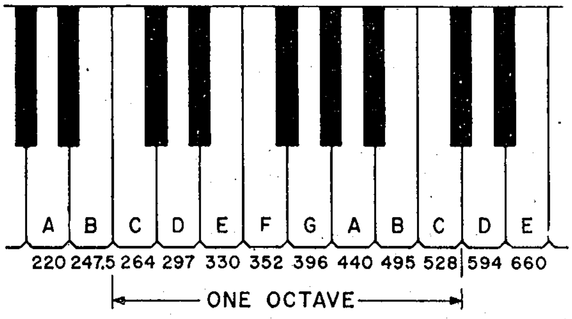
Fig. 5. Quality of Sound.

This wave is produced by the fundamental wave, at the top left, plus its third and fifth harmonics. The wave at the top is one cycle of a 100 c/s wave having an amplitude of ten units. To this is added its third harmonic, which would be 300 c/s having an amplitude of five units, and the fifth harmonic, which would be 500 c/s, having an amplitude of two and a half units, to obtain the resultant wave form shown at the bottom. Conversely, any irregular periodically recurrent wave can be analysed into its fundamental frequency and harmonics. It is the harmonics, or overtones as they are called in music, that give the characteristic quality to a particular note. For instance, middle "C" played on the piano sounds different from middle "C" on a violin, although they are both the same note, and both produced by strings. This difference is caused entirely by the amplitude and number of the harmonics produced at the same time the fundamental is produced. This can also be said of sounds which are not musical tones. For instance, one person's voice is different from another person's voice because of overtones. The voice is produced by the vibration of vocal chords and their resonance in the mouth and throat cavities. No two people are built alike, consequently their voices do not sound alike. Some people have long vocal cords and consequently a longer vibratory column. This makes good bass singers. Other people have short vocal cords. This makes good tenor singers.

On the right side of figure 5 are shown four graphs of sound waves producing the note, middle "C". The top graph is a sound wave of a cello organ pipe (an open flue pipe) when playing middle "C" The second chart is a trombone organ pipe (a reed pipe) when producing middle "C". The third is middle "C" as played on a piano, and the one on the bottom is the syllable "ah", when being sung by a soloist at middle "C". All of these are middle "C", 264 c/s. It can be seen that the characteristic outlines are different in each of the graphs. They are different because the different instruments or various organ pipes superimpose upon the fundamental frequency a different set of harmonics, and thereby endow the note with an individual characteristic.

THE MUSICAL SCALE

Figure 6 shows a small portion of a piano keyboard, and the section covered by one octave is indicated starting with middle "C" and going up. American Standard pitch for middle "C" is 264 c/s. For each octave the frequency doubles and consequently the frequency of "C" one octave above middle "C" would be 528 c/s. Inasmuch as the scale is divided into twelve half-tones, each half-tone will have a difference in frequency from the one adjacent to it by the 12th root of 2. The twelth (12th) root of 2 is 1.059, so if the frequency of any note is multiplied by 1.059, the frequency of the half-tone above it will be obtained. That is, one half-tone above middle "C" will be "C" sharp and would be played on the black key between "C" and "D". It will be noticed that there is no black key between "E" and "F" and again no black key between "B" and "C". However, there is still only a half-tone between these keys. This arrangement is necessary in order that music may be played in various keys.



THE FREQUENCY DOUBLES EACH OCTAVE.
THE FREQUENCY INCREASES BY THE TWELFTH ROOT OF TWO (1.059) FOR EACH HALF TONE.

MUSICAL CHORDS ARE COMBINATIONS OF NOTES WHICH ARE PLEASING TO THE EAR, SUCH AS C, E, AND G.

Fig. 6. Musical Scales.

As has already been seen, each one of these keys, when depressed, not only produces the fundamental frequency but also an array of overtones or harmonic frequencies. It is because of these overtones primarily that musical chords can be produced. A musical chord is defined as a combination of notes which are pleasing to the ear. For instance, the notes, C, E and G, when struck in unison, produce a pleasing effect. Also the notes F, A and C produce a pleasing musical chord. If D, E, and F, are struck, only a jarring discord would be produced which would not be pleasing to the ear. To be pleasing to the ear, the frequencies of the notes struck must conform to each other in a definite ratio, that is, the frequencies of C, E, and G are in the proportion of 4:5:6. The frequencies of F, A, and C are also in this same proportion. Other proportions can be found which will give other types of chords.

$$\left.\begin{matrix}C:E:G\\F:A:C\end{matrix}\right\}4:5:6$$

$$\begin{array}{r|lll} & C & E & G\\ 66 & 264 & 330 & 396\\ \hline & 4 & 5 & 6\end{array}\qquad\begin{array}{r|lll} & F & A & C\\ 88 & 352 & 440 & 528\\ \hline & 4 & 5 & 6\end{array}$$

NOTE	FUNDAMENTAL	HARMONICS							
		2 ND.	3 RD.	4 TH.	5 TH.	6 TH.	7 TH.	8 TH.	9 TH.
C	264	528	792	1056	1320	1584	1848	2112	2376
E	330	660	990	1320	1650	1980	2310	2640	2970
G	396	792	1188	1584	1980	2376	2772	3168	3564

Fig. 7. Musical Chords.

MUSICAL CHORDS

Figure 7 shows the chords of C, E, G and F, A, C, having a ratio of 4:5:6. Any of these notes might be replaced by the corresponding note an octave above or below, or two octaves above or below, and so on, without disturbing the harmony of the chords. The frequency of middle "C" is 264, the "E" above is 330, and "G" above is 396 c/s. Dividing these three frequencies by 66 a ratio of 4:5:6 is indicated. For the chord F, A, C, "F" is 352, "A" 440, and "C" is 528 c/s, dividing by 88, the ratio of 4:5:6 is again obtained. In the chart at the bottom of figure 7 are listed the fundamental frequencies for "C", "E", and "G", plus their harmonics up to and including the ninth harmonic.

On this chart is pointed out some of the reasons why these notes make a pleasing combination. It will be noticed that the third harmonic of "C" is the same as the second harmonic of "G". The fifth harmonic of "C" is the same as the fourth harmonic of "E". The sixth harmonic of "E" is the same as the fifth harmonic of "G", and the ninth harmonic of "C" is the same as the sixth harmonic of "G". These are only some of the reasons why this is a pleasing chord. The beat between these various harmonics can be found by subtracting the harmonic frequencies and if a number of these subtractions are made, it will be found that many of the beat notes between the harmonics correspond to other harmonics of the fundamentals.

OVERTONE RANGE OF INSTRUMENTS

Figure 8 shows the frequency range of various instruments. One would hardly expect that the sound of a snare drum would go higher in frequency than the highest fundamental note of a violin, but this is the case. A snare drum is a percussion type of instrument. That is, it is struck in order to produce sound. Across the drum head are stretched several strings which rattle on the surface of the head. It is this rattling which produces the high frequencies and carries the range of the snare drum so high. Also, the sound of the bass drum goes much higher than would be expected. When it is struck it sounds only a dull thud though it has a fundamental range to over 1,000 c/s. However, it has high harmonics which go almost as high in frequency as the ordinary piano scale. It will be seen that the fundamental range of the female voice covers from approximately 170 c/s to less than 2000 c/s, and the range of the male voice is from approximately 100 c/s to just over 500 c/s. They both have harmonics many times higher than these frequencies. The sound produced by hand clapping (which although not being considered as a musical noise, must be considered in the art of sound reproduction) runs from approximately 100 c/s to 16,000 c/s which is beyond the range of audibility for many persons. The sound of footsteps is from approximately 80 c/s to about 1,500 c/s.

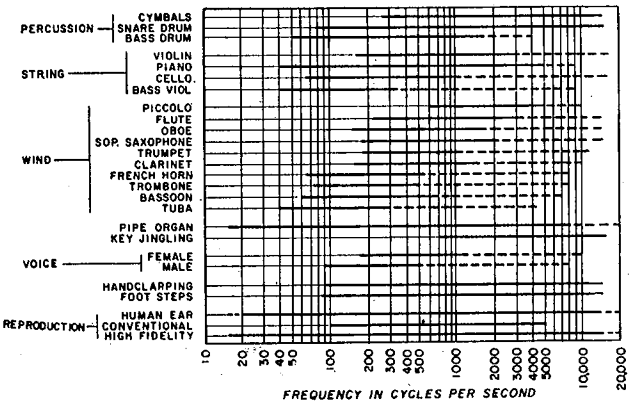


Fig. 8. The Frequency Range of Musical Instruments.

As can be noted, practically no musical instrument produces fundamental frequencies above 5,000 c/s, but their tonal ranges are often several times extended by the higher order harmonics.

Near the bottom of figure 8 is a line representing the range of the human ear. The average person can usually hear from about 20 c/s through approximately 15,000 c/s though this usually is reduced as the age of the person increases. Some people have even a greater range than the average just given, being able to hear as the dotted extensions of this line indicate, from about 15 c/s to 20,000 c/s or over.

(Continued on page 84).

VIBRATORS

mercial vibrator is not just
switch, but a precision unit
uracy of between 1% and
f the various operations oc-
and built to maintain these
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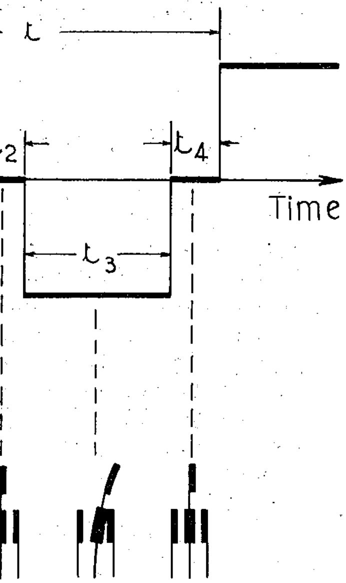

Fig. 1.

vibrator. In the design of such transformers care has to be taken to ensure that the iron core will not saturate when the maximum operating voltage is applied, which means that relatively low flux densities are required at normal voltage. Operation of a vibrator outside its rated limits or under incorrect circuit conditions will invariably lead to premature failure. The following notes are intended to aid in the design of the various circuit components.

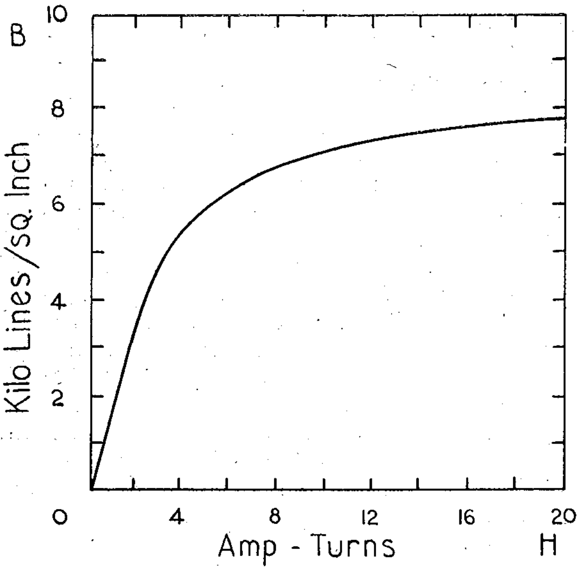

Fig. 2.

VIBRATOR CHARACTERISTICS

A complete knowledge of vibrator characteristics is required before starting to design any of the electrical components to be used in the power supply circuit. The more important of these characteristics are the time efficiency and the operating frequency, the latter being fixed on all standard vibrators at 115 c/s, which is the frequency generally used overseas.

The time efficiency of a vibrator refers to the time that the interrupter contacts are held in contact during each cycle. This varies from 70% to 90% of the period of the cycle, depending on the type of vibrator. A graphic illustration of this is given in Fig. 1, which is the CRO pattern obtained when a vibrator is fed into a centre-tapped resistor, and which also indicates the

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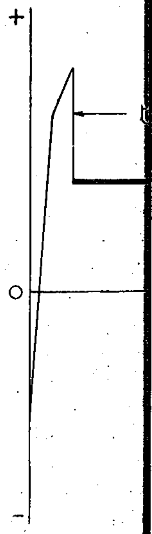

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relative positions of the contacts during one cycle.

One pair of contacts is closed for the time "t1", and the other for "t3". For these times and the total time for one cycle ("t"), the time efficiency may be determined from the expression:

$$\text{Time efficiency} = \frac{t1 + t3}{t} \times 100\%$$

The above characteristics apply to all vibrators regardless of whether they are synchronous or non-synchronous types. However, synchronous vibrators differ from non-synchronous types in that they have a second set of contacts identical in construction to the interrupter contacts, but adjusted to close after and open before those contacts.

TRANSFORMERS

With a knowledge of the operating conditions and vibrator characteristics, it is possible to design a transformer giving the required voltage gain and also matching the vibrator to be used. A timing capacitor is required in the circuit, the purpose of which is described later. It can be shown that the value of timing capacitor required depends directly on the magnitude of the magnetisation current of the transformer. It is desirable therefore that this current be kept as constant as possible over the supply voltage operating range, to ensure satisfactory tuning conditions.

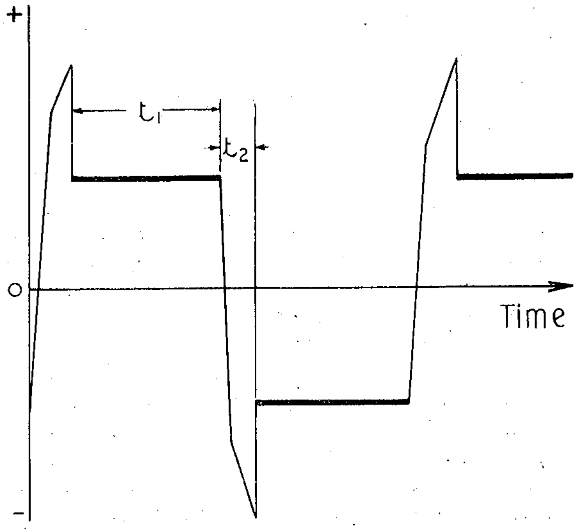

Fig. 3.

It is only possible to approach this ideal by operating on a straight line section of the BH curve of the transformer iron used. (See Fig. 2.) There are two such sections, one at low flux densities, and the other as the iron is approaching saturation. The former results in uneconomic construction and the latter in severe loss of efficiency. It is therefore customary to compromise with a value of between 30,000 and 35,000 lines per square inch for the maximum input voltage.

Having determined the flux density range of the transformer, the remaining factors affecting design are the number of primary turns and the cross-sectional area of the core. It is customary to select a figure of 4 or 5 turns per volt for the windings to reduce the leakage inductance.

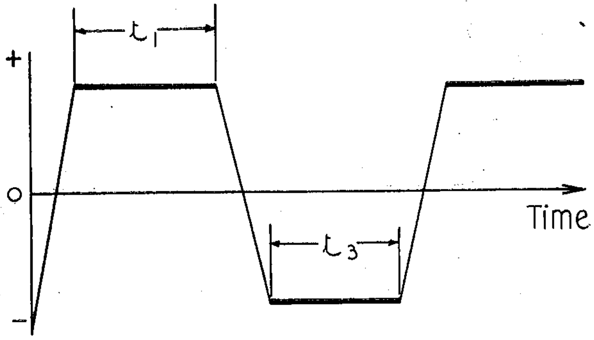

Fig. 4.

As a result of this and the low flux density required, it is usual to use a larger core size in a vibrator transformer than would be required in a normal AC transformer.

TIMING CAPACITOR

A buffer or timing capacitor is required in a vibrator circuit as a protection against the high transient voltages induced when pulsating DC is applied to an inductive load. Without a timing capacitor the waveform produced when a vibrator is used with a transformer is as shown in Fig. 3.

Here it can be seen that there are high transient peaks developed as soon as each pair of contacts is opened. In practice these peaks are many times the value of the input voltage, and thus impose undue strain on the transformer insulation and the vibrator contacts. The addition of a capacitor completes a damped oscillatory circuit with the leakage inductance of the transformer. If the correct value of capacitor is used,

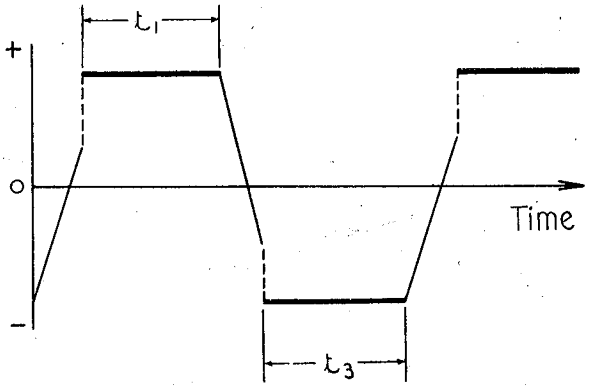

Fig. 5.

the waveform will be as shown in Fig. 4.

Here the voltages in the transformer are controlled so that the windings or vibrator contacts are never subjected to voltages in excess of the input voltage. It can also be seen from this figure that the time efficiency of the vibrator will directly affect the value of timing capacitor. Vibrators with short transit times, i.e., short no-contact periods, require a lower value of capacitor than those with long transit times, if used with the same transformer.

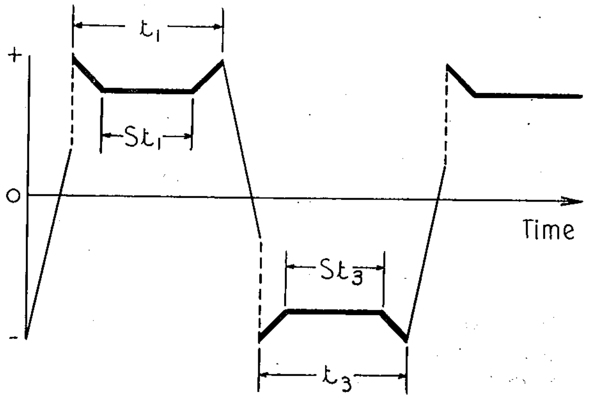

Fig. 6.

PRACTICAL APPLICATION OF TIMING CAPACITOR

For correct operation, the timing capacitor should be applied across the whole of the primary winding of the transformer, but in lower voltage power supplies this capacitor would be of prohibitive size because of the value required. In such cases it is customary to connect a capacitor of a suitable value across the secondary winding, depending on the transformer characteristics to reflect this capacitance to the primary. This method is only possible with units operating from 12V. or less. With voltages higher than 12V., gas ionisation is possible, leading to arcing between contacts and subsequent failure of the vibrator.

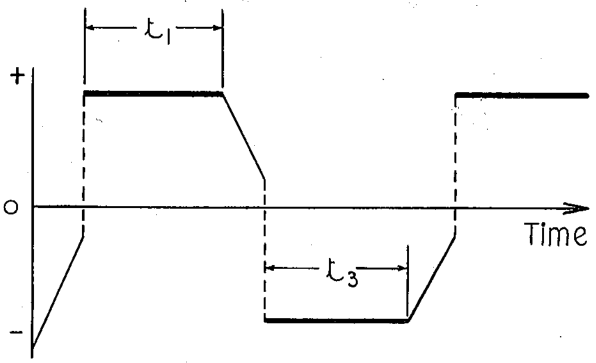

Fig. 7.

With operating voltage higher than 12V., it is possible to have part of the required capacitance connected to the secondary windings and part on the primary. This will ensure instantaneous suppression of any transient voltage present. The percentage of the total capacitance to be used on the primary winding must be determined experimentally by observing the degree of arc suppression obtained.

The timing capacitor value may be calculated from transformer and vibrator characteristics, but because of production variations in these items, it is usually more satisfactory to find the correct value experimentally. A capacitor of slightly larger value than that giving a perfect waveform should be used. This will allow for the wear that takes place in the vibrator contacts during their life, which results in a decrease in time efficiency, i.e., an increase in the transit times.

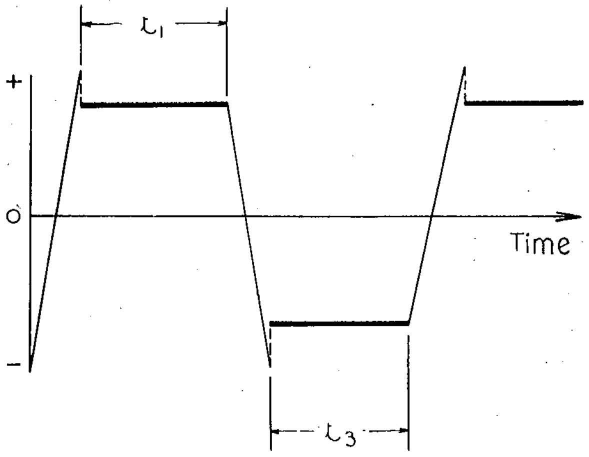

Fig. 8.

The correct wave form for a new interrupter-type vibrator is illustrated in Fig. 5. These waveform considerations apply equally to synchronous as well as non-synchronous vibrators. The correct waveform for a synchronous vibrator is shown in Fig. 6, where the closure times for the second set of contacts are designated St1 and St3.

The effects of using incorrect values of timing capacitor are illustrated in Figs. 7 and 8. Fig. 7 shows the wave form obtained when the timing capacitor value is too large, resulting in under-closure. Fig. 8 shows the dangerous effect of over-closure, resulting from the use of too low a value of timing capacitor. Here it can be seen that there is a tendency for peaks to form, which could cause arcing betwen contacts.

ACKNOWLEDGEMENT

This article is based on information kindly made available by Manufacturer's Special Products Pty. Ltd.

A RATIO DETECTOR FOR F-M BROADCAST RECEIVERS*

INTRODUCTION

The B.B.C. "frequency modulation" broadcasting network was introduced to provide listeners with better reception. The transmitted frequencies lie in the range 87.5—100 Mc/s. In general, receivers will be of the superheterodyne type and will probably have an intermediate frequency of 10.7 Mc/s.

The overall performance of the receiver depends to a large extent on the properties of the detector (discriminator) stage. Various circuit arrangements can be used for this, the one dealt with here being known as the Ratio Detector; it combines two functions, namely that of detection and that of amplitude limiting. The purpose of the latter function is to remove any undesirable amplitude modulation superimposed on the input signal. This amplitude modulation may arise for a number of reasons including "ignition interference" which, being random, gives rise to non-synchronous A-M. The F-M signal may also be distorted as it passes through the receiver if the frequency response characteristics of the amplifiers are too selective (i.e. too narrow); this type of distortion is said to be synchronous. Other detector systems, such as the Foster-Seeley discriminator, must be preceded by a suitable amplitude limiting circuit. This limiting usually takes place at the grid of the amplifier valve feeding the detector. Since no limiting is necessary when using a ratio detector the input level at the grid of this driver stage can be much lower. In practice this means that fewer stages are required in the I-F amplifier.

BASIC PRINCIPLES OF THE RATIO DETECTOR

F-M Detection

A fundamental property of a transformer having its primary and secondary windings tuned to the same frequency is that the secondary voltage is 90° out of phase with that appearing across the primary. A transformer having one end of the primary connected to the centre tap of the secondary will result in the voltages represented by the vectors e_1 and e_2 in figure 1 being measured between the free end of the primary and the remote ends of the secondary in turn, while the vectors e_p and e_s represent the primary and secondary voltages. Small variations in the frequency of the applied signal will not, to a first approximation, affect the amplitude of the primary voltage; however, the above 90° phase relationship will be disturbed so that the secondary voltage may then be represented by vectors such as AB, the points A and B lying on circles as shown.

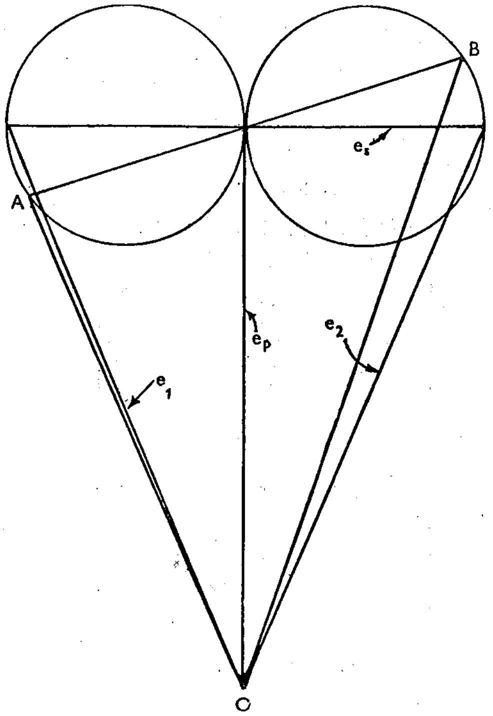

FIG. 1

The two vectors OA and OB, which are amplitude modulated, have now been derived from the frequency modulated input signal such that the relative magnitudes of OA and OB depend on the instantaneous frequency, i.e. the difference between their magnitudes is directly proportional to the instantaneous frequency. The voltages represented by OA and OB may therefore be applied to amplitude detectors such as diodes, after which the difference between the rectified voltages, being proportional to the instantaneous frequency of the input signal, forms the desired audio output signal.

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* Reprinted with acknowledgment to G.E.C.

A-M Rejection

The basic circuit of the ratio detector is shown in figure 2. To enable the ratio detector to reject any A-M present in the input signal it is necessary that (a) the sum of the rectified voltages E1 + E2 is kept at a constant value and (b) the ratio E1/E2 of the rectified voltages is always equal to the ratio OA/OB of the frequency sensitive I-F voltages; as a result the difference E1—E2 always represents the audio output due to F-M and is independent of any A-M component in the input signal. The sum of the rectified voltages can be held constant by connecting either a battery or a large storage capacitor across the output terminals. In practice the capacitor is preferred because it stabilises the output at a value determined by the mean input signal; thus it automatically provides compensation for any long term changes in input level. When a battery is used its voltage must be carefully chosen; if it is too high a threshold level of input signal exists below which the detector no longer functions properly, whereas if it is too low the detector becomes inefficient.

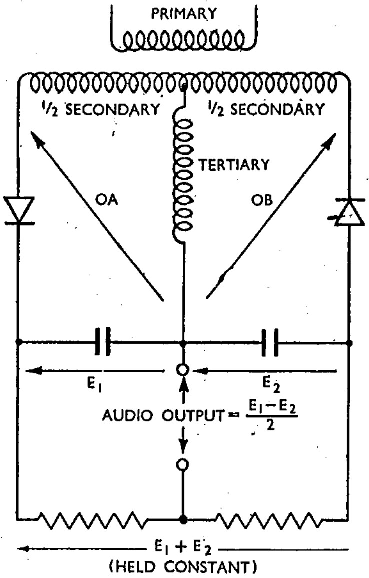


FIG. 2
Basic Circuit of Ratio Detector.

The operation of the detector may be seen from the following simplified analysis. If the detection efficiencies of the diodes in the two halves of the circuit are equal then E1 is proportional to OA, and E2 is proportional to OB, which may be written

$$E_1 = K\,OA \quad \text{where K is a suitable constant}$$

$$E_2 = K\,OB$$

$$\therefore\ E_2 = E_1 \frac{OA}{OB} \quad \text{.......................................(1)}$$

The sum of the rectified voltages is held constant

$$E_1 + E_2 = V$$

Substituting for E_2 from (1)

$$E_1\left(1 + \frac{OB}{OA}\right) = V = E_1 \frac{OA + OB}{OA}$$

$$\text{or } E_1 = \frac{V \times OA}{OA + OB} \quad \text{.................................(2)}$$

The required audio output is $E_1 - E_2$ which may be written

$$E_1 - E_2 = E_1 - E_1 \frac{OB}{OA}$$

$$= E_1 \frac{(OA - OB)}{OA}$$

Substituting for E_1 from (2) we obtain

$$E_1 - E_2 = \frac{OA.\,V}{OA + OB} \times \frac{OA - OB}{OA}$$

$$= V \frac{OA - OB}{OA + OB} \quad \text{.................................(3)}$$

The effect of the undesirable A-M is to increase or decrease OA and OB in the same ratio i.e. make them equal to m.OA and m.OB, where m is a suitable factor representing the magnitude of the A-M.

Then $E_1 - E_2$ becomes

$$V \frac{m.OA - m.OB}{m.OA + m.OB}$$

$$= V \frac{OA - OB}{OA + OB} \quad \text{.......................................(4)}$$

In other words the output $E_1 - E_2$ remains, as before, the output due to F-M and has not been affected by the A-M present in the input signal.

The action of the large capacitor (or battery) used to stabilise the sum of the rectified voltages may be visualised as follows. When the A-M in the input signal tends to drive the diodes harder the current due to the A-M component flows into the battery. When the A-M tends to reduce the diode drive, the battery supplies current to the load resistors to keep the voltage across them at the average level. When the A-M in the input signal instantaneously reduces the diode drive to too low a value the stabilising voltage effectively prevents the diodes from conducting. When this occurs the detection process temporarily ceases, and the percentage modulation causing this effect is termed the downward A-M capability of the detector. No similar limit to the amount by which the signal amplitude may be increased exists. Thus it is seen that the ratio detector does not provide perfect limiting. In addition expressions (3) and (4) give an over-simplified picture which does not completely hold in practical detector circuits.

A.V.C.

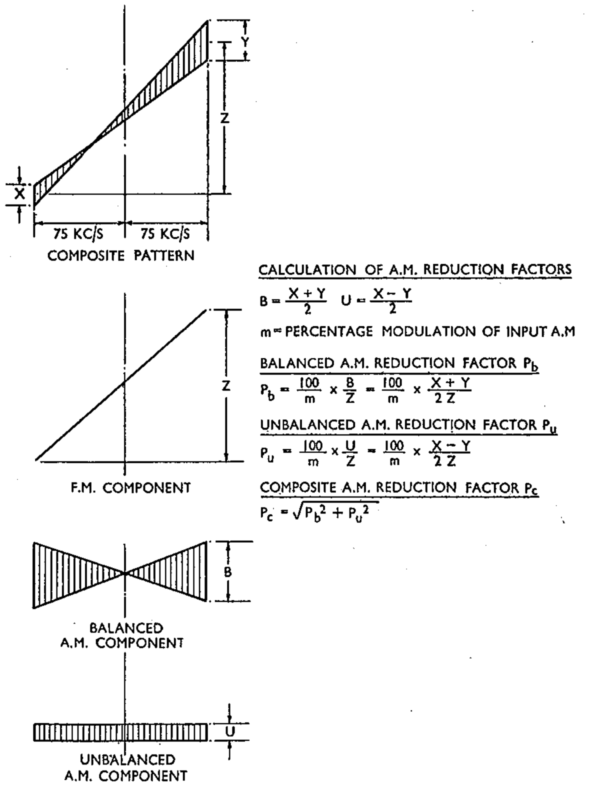
If the sum of the rectified voltages is stabilised by a large capacitor, it will adjust itself to correspond to the average input signal level, and is available for use as an A.V.C. voltage. In the balanced circuit considered, which has the centre point of the diode circuit load resistor grounded, half the sum of the rectified voltages is available for A.V.C. use. However, in the unbalanced circuit one end of the load resistor may be grounded, whereupon the full sum of the rectified voltages may be used for A.V.C. purposes.

IMPORTANT PARAMETERS AND THEIR MEASUREMENT

The important properties of the ratio detector are its linearity, sensitivity and A-M rejection performance; the last property is defined by two factors, namely the downward A-M capability as defined above, and an A-M reduction factor (P). The A-M reduction factor expresses the ratio between the audio output due to A-M and that due to F-M, both being corrected to correspond to an input signal simultaneously amplitude and frequency modulated to a depth of 100% (a frequency deviation of ± 75 Kc/s is arbitrarily termed 100% frequency modulation); thus an A-M reduction factor of zero represents perfect A-M suppression, while a value of unity represents no A-M rejection whatsoever. The A-M rejection can be expressed in decibels by calculating $20 \log_{10} \frac{(1)}{(P)}$

The measurement of sensitivity and linearity are straightforward, the latter being done either by using a wave analyser, or by inspection of the detector dynamic response characteristic on a C.R.O. The measurement of the A-M rejection parameters may be carried out by a number of methods, the preferred ones being described below.

A simultaneously frequency and amplitude modulated signal must be used to measure the A-M reduction factor if its correct value is to be obtained. In one commonly used method the input signal is consecutively either frequency or amplitude modulated, the A-M reduction factor then being defined as the ratio of the resulting audio output voltages; unfortunately this method yields optimistic results since the A-M rejection is really only being measured at the detector centre frequency where it is theoretically perfect. In the preferred method the detector dynamic



CALCULATION OF A.M. REDUCTION FACTORS

$B = \frac{X+Y}{2} \quad U = \frac{X-Y}{2}$

m = PERCENTAGE MODULATION OF INPUT A.M

BALANCED A.M. REDUCTION FACTOR P_b

$P_b = \frac{100}{m} \times \frac{B}{Z} = \frac{100}{m} \times \frac{X+Y}{2Z}$

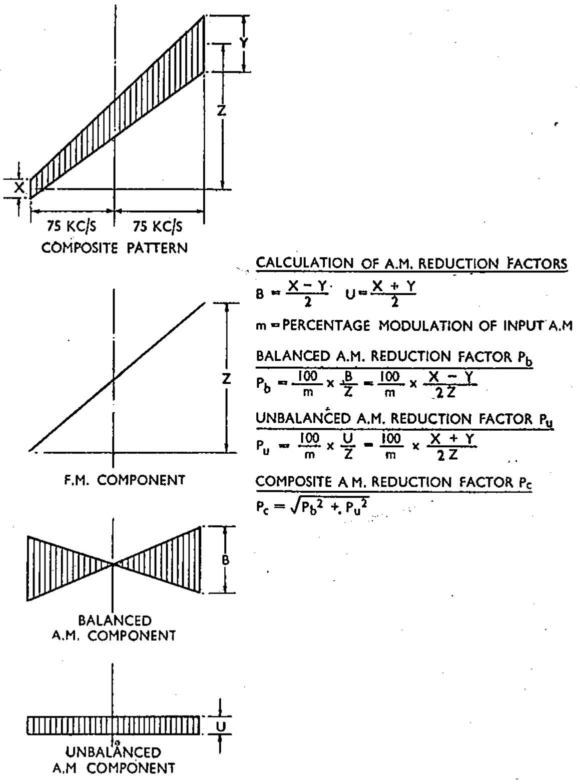
UNBALANCED A.M. REDUCTION FACTOR P_u

$P_u = \frac{100}{m} \times \frac{U}{Z} = \frac{100}{m} \times \frac{X-Y}{2Z}$

COMPOSITE A.M. REDUCTION FACTOR P_c

$P_c = \sqrt{P_b^2 + P_u^2}$

Fig. 3a. Dynamic Response Pattern Calculation of A-M reduction factors for the case of crossover on pattern.



CALCULATION OF A.M. REDUCTION FACTORS

$B = \frac{X-Y}{2} \quad U = \frac{X+Y}{2}$

m = PERCENTAGE MODULATION OF INPUT A.M

BALANCED A.M. REDUCTION FACTOR P_b

$P_b = \frac{100}{m} \times \frac{B}{Z} = \frac{100}{m} \times \frac{X-Y}{2Z}$

UNBALANCED A.M. REDUCTION FACTOR P_u

$P_u = \frac{100}{m} \times \frac{U}{Z} = \frac{100}{m} \times \frac{X+Y}{2Z}$

COMPOSITE A M. REDUCTION FACTOR P_c

$P_c = \sqrt{P_b^2 + P_u^2}$

Fig. 3b. Dynamic Response Pattern Calculation of A-M reduction factors for the case of crossover off pattern.

characteristic will have one of the forms shown in figure 3; the division of the reduction factor into balanced and unbalanced parts as shown is convenient not only for the calculation of the overall reduction factor but also indicates what adjustments are required to improve the detector performance.

The downward A-M capability may be measured by increasing the A-M of a simultaneously frequency and amplitude modulated signal until the residual A-M begins to increase very rapidly, whereupon this level of A-M gives the above factor directly.

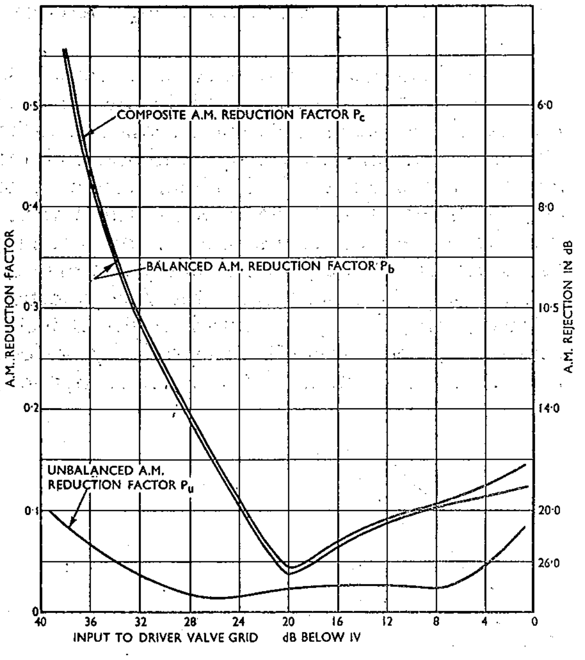


Fig. 4.

PRACTICAL CIRCUIT

The A-M rejection performance of a practical ratio detector may be seen from figure 4, which gives the relation between A-M reduction factor and input signal level. Many factors combine to alter the position and exact form of the curves and a compromise has to be achieved in the design process, which is therefore largely empirical. Some of the factors involved are (a) the ration of the half secondary to tertiary voltages; (b) the coupling between the primary and secondary; (c) the magnification factors (Q) and L/C ratios of the primary and secondary tuned circuits; (d) the fraction of the rectified voltages to be stabilised; (e) the characteristics of the diodes, and (f) the values of the load resistors. Most of these parameters are interdependent. Subject to certain limitations, the unbalanced residual A-M may be optimised by adjusting the resistors R1 and R2 in figure 5, while the balanced residual A-M is minimised by adjustments in the primary to secondary coupling. To obtain the best performance close attention must be paid to tolerances during the production of ratio detector transformers, and no changes in the type of diode used are permissible without making subsequent adjustments to the transformer and circuit.

To derive an undistorted audio output from the detector it is necessary that the carrier frequency of the received signal corresponds to the detector centre frequency. Temperature changes can disturb this condition, so that it is beneficial to apply temperature compensation catering for drifts due to the linear circuit components; compensation for the second order drift caused by changes in diode parameters cannot be simply applied since neither the magnitude nor sense of this drift can be predicted for any particular pair of diodes.

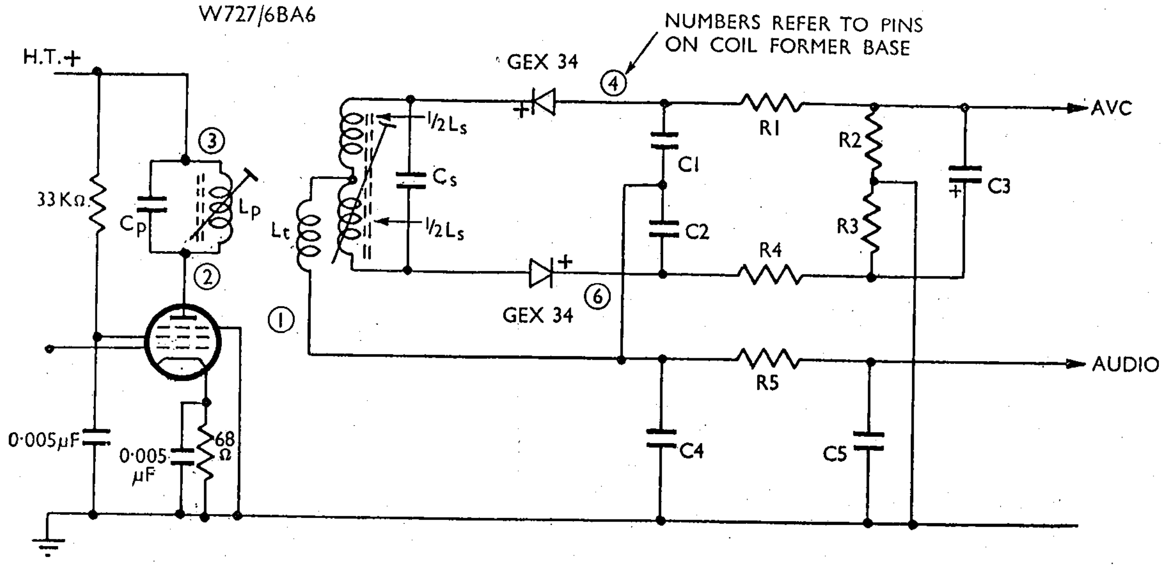
The use of germanium diodes in the ratio detector leads to the following advantages:

1. The small size of germanium diodes enables them to be located inside the detector transformer screening can, thus reducing the possibility of stray coupling with other parts of the receiver.
2. No necessity exists for heater supplies and the associated leads; this is important in battery receivers where power consumption must be kept to a minimum. Also there is less likelihood of hum pickup.
3. The small effective shunt capacitance of the germanium diode and the absence of valve-holder capacitance reduce circuits strays.

A ratio detector circuit and transformer suitable for use with germanium diodes have been evolved, the circuit diagram being given in figure 5. The transformer is wound on a Salford Electrical Instruments former, type 61B, fitted with two S36B.ME iron dust cores (see figure 6). It is screened by a standard ⅞ in. square aluminimum can, which also encloses the tuning capacitors and the diodes. The values given are for a frequency of 10.7 Mc/s. The winding data is as follows:—

Primary:	35 turns 36 S.W.G. enamelled and single silk covered copper wire, close wound.
Secondary:	16 + 16 turns 28 S.W.G. enamelled and single silk covered copper wire. It is BIFILAR and close wound (see below).
Tertiary:	9½ turns 36 S.W.G. enamelled and single silk covered copper wire, close wound over paper interlay approximately 0.002 in. thick, placed over end of primary remote from secondary.

All windings to be in the same direction.



R1 1KΩ approx.
R2 6.8KΩ
R3 6.8KΩ
R4 1KΩ approx.
R5 22KΩ

C1 330μμF. cer.
C2 330μμF. cer.
C3 8μF.
C4 330μμF. cer.
C5 0.002μF.

Cp 10μμF. cer. N750
Cs 39μμF. cer. N330
OR 18μμF. silver
mica + 22μμF.
cer. N750

Fig. 5.

Ratio Detector Circuit Diagram.

The secondary consists of two coils in a bifilar arrangement; that is they are interwound as shown in figure 6, where they are represented by the full and dotted lines respectively. The use of a bifilar winding ensures that the two halves of the secondary are well balanced and closely coupled.

It is convenient to wind the secondary before the primary and tertiary, the procedure being as follows. A length (about two yards) of 28 S.W.G. wire is folded in the middle, one (double) end is temporarily secured to the coil former base, and all the wire is wound onto the former. The coil is then fixed in position by means of a suitable adhesive such as polystyrene varnish. The distance from the coil former base to the bottom end of the secondary is adjusted by carefully removing turns; some turns are then removed from the top end of the winding until the correct number of turns is obtained. The upper end A1 of coil A is then soldered to the lower end B2 of coil B at a point P midway between A1 and B2, as shown in figure 6.

The spacing between the primary and secondary must be adjusted accurately and it may therefore be found convenient to wind the primary on a paper sleeve which is just capable

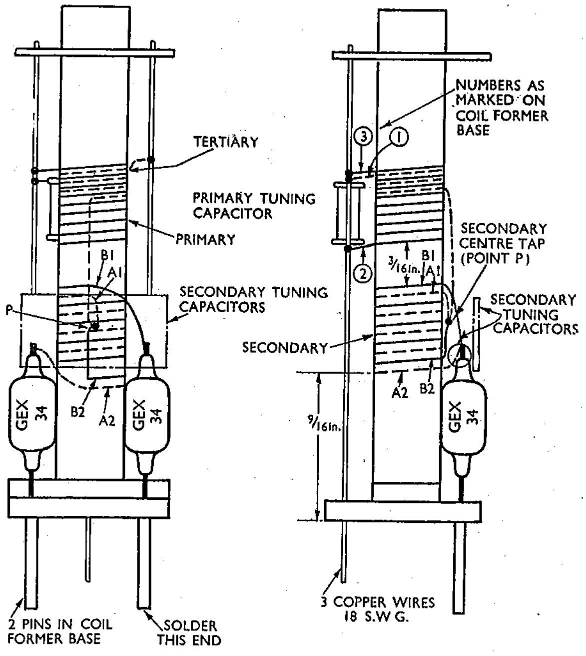


Fig. 6.

Ratio Detector Transformer Details.

of sliding along the coil former; the coupling between the windings may then be adjusted when they have been completed. Both ends of the primary and the top end of the tertiary are soldered to the 18 S.W.G. copper wires as shown in figure 6; the numbers indicated on figures 5 and 6 refer to those stamped on the coil former base. The lower end of the tertiary is soldered to the point P mentioned above.

The diodes are anchored in the coil former base by soldering them to the eyelets; during this process care must be taken not to overheat and so damage the diodes and the use of a thermal shunt such as a pair of pliers is recommended. It is preferable to replace the eyelets in holes 4 and 6 in the coil former base by hollow pins; the diode leads may then be passed through the pins and soldered at the ends remote from the coil former base, with the result that there is less likelihood of overheating the diodes.

To align the ratio detector, a frequency modulated 10.7 Mc/s signal is applied to the grid of the driver valve, whereupon the primary tuning core is adjusted to give maximum reading on a high impedance voltmeter connected across R2 (figure 5), whereupon the secondary core is adjusted to give minimum reading on this meter. The meter is then connected between the audio output terminal and earth, and the secondary is tuned for zero reading on the meter. Next, with the meter again connected across R2, the primary is retuned for maximum reading on the meter. The detector is then arranged to give optimum A-M rejection by applying an amplitude modulated signal and adjusting R_1 or R_4 (figure 5) for minimum output due to A-M.

A convenient way of applying A-M for this adjustment is to amplitude modulate the I-F signal generator or to inject some mains frequency hum from the heater supply to the cathode of the last I-F amplifier valve. It may subsequently be found that the D.C. balance of the circuit has been disturbed and the above tuning procedure is then repeated to regain balance as indicated by zero direct voltage at the audio output terminals of the detector. Thus it is often necessary to carry out the I-F alignment of the receiver twice.

EDITOR'S NOTE

The ABC broadcasts from experimental F-M stations in Sydney, Melbourne, Adelaide and Brisbane. The stations are open from 11 a.m. — 11 p.m., and broadcast either the national or the light programme according to the type of programme material available. The frequencies are as follows:

Sydney	92.1 Mc/s	Adelaide	97.3 Mc/s
Melbourne	91.1 Mc/s	Brisbane	91.1 Mc/s

The S.E.I. type 61B coil former mentioned in the article may be replaced by almost any standard former of similar construction with a winding former diameter of approximately 3/8" and a 7/8" screening can. The numbering of the base connections may, of course, be different where a substitution is made.

(Continued from page 75)

Because this discussion is on the subject of high fidelity a line has been added showing the average range of the conventional amplifier which has a frequency response of from approximately 100 c/s to 5,000 c/s. Often, in the conventional amplifier, even the response at 5,000 c/s is extremely low. At the very bottom of the chart is shown a line representing the range of a good high fidelity amplifier. This as can be seen, runs from 15 c/s to 20,000 c/s. Good high fidelity response can still be obtained, however, if the high frequency response goes out to only 15,000 c/s. It can be easily seen why such a range is necessary if the upper portion of the chart is again perused, particularly the line showing the range of the pipe organ. Naturally, if the full high fidelity range is not available, certain notes of the organ and certain other instruments will definitely be lost. This reasoning also applies to the high frequencies.