RADIOTRONICS

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The Establishment of Quality Standards by Subjective Assessment

by T. Somerville *

The maintenance and improvement of quality standards in any electro-acoustic system, whether it be broadcasting or sound recording, requires constant supervision, a comprehensive measuring technique, and a reliable system of subjective assessment. There has been considerable development in measuring techniques, but there is so far no established system of subjective tests. Yet it is the subjective assessment which provides the final verdict on the capabilities of a system.

Listening tests of one type or another are always carried out, but there are pitfalls in subjective testing. It may therefore be of advantage to outline methods used in the B.B.C. as a guide to other workers.

No doubt it will be generally agreed that reproduction of sound by electro-acoustic means can never be an exact replica of the original. This is so for a number of reasons. First and most important, most recording and broadcasting systems are not stereophonic; it is not economic to employ more than one channel between the listener and the studio. There are also inevitable distortions of the signal in every part of the system, so that it is quite impossible to reproduce the original exactly.

As the original sound cannot be duplicated, two fundamental decisions have to be made. The first decision by most organisations is that every attempt should be made to reproduce the original as closely as possible, because only in this way can a standard, reasonably unaffected by matters of opinion, be obtained.

As the listening room acoustics also have a profound effect on the results, a decision is necessary on the standard listening conditions. It is often argued that listening rooms should be as "dead" as possible so that their acoustics cannot have an appreciable effect on the programme. In this connection it should not be forgotten that the polar diagram of the loudspeaker also affects the subjective reverberation time of the room. Some organisations make studio listening cubicles dead; they perhaps do not realise that monitoring of the programme may be misleading because the ordinary listener does not listen under very dead conditions. Experiments have shown that in Britain the reverberation time of the average living room is of the order of half a second, and for this reason most of the B.B.C.'s critical subjective tests are carried out in listening rooms having a reverberation time of this order. These rooms are furnished as far as possible in the manner of living rooms.

In carrying out subjective tests it is usual to proceed by a series of successive assessments, each part of the chain being dealt with in turn. Such an investigation becomes very elaborate and must be conducted with care if reliable results are to be obtained. It should be noted that it is quite possible for one element of the chain to compensate for faults in another, so that improvement in one part sometimes results in an unexpected deterioration in over-all performance. The best way to illustrate some of the difficulties is to describe some experiments conducted by the B.B.C.

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* British Broadcasting Corporation, London.
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After the war it became necessary to select a new loudspeaker for monitoring B.B.C. programmes. Manufacturers were therefore asked to submit samples of high-grade loudspeakers for tests. The observers taking part in the subjective assessments were not given the results of these tests until their opinions had been recorded.

The first subjective comparisons were made in a listening room on the B.B.C. programmes available. It was found that although the majority of the loudspeakers could be rejected as unsuitable, some appeared to be satisfactory only on certain types of programme. Furthermore, the order of preference was found to vary with the programme material. It was soon realised that a decision could not be made unless comparison with the original programme was possible, and so the next series of tests was carried out at a studio centre.

Here again the results were inconclusive for the same reason, and it was further noted that wider range loudspeakers were normally disliked because of harsh quality. Careful listening tests conducted in the studio led to the conclusion that the acoustics were indifferent. The tonal quality was harsh, and in loud passages the weaker instruments became inaudible.

To discover whether or not the anomalies were due to poor acoustics the whole experiment was transformed to a good orchestral studio, whereupon it was quickly found that reliable comparisons could be made resulting in the rejection of all but three of the loudspeakers under test. The three remaining loudspeakers were a wide-range unit of American origin, and two single-cone units of an early type (A and B) both made by the same manufacturer. It is interesting that this was the first occasion on which a wide-range loudspeaker gave pleasing results.

At this juncture it was realised that the loudspeaker employed in the studio cubicle was type B. Lest this should be affecting the choice, type B was replaced by type A with quite surprising results. The microphone had to be moved much further from the orchestra and raised higher to find the best position, which then gave much superior results on the wide-range loudspeaker and caused the rejection of the single-cone unit type B. To complete the experiment the orchestra was "balanced" on a wide-range unit and gave satisfactory results on the single-cone unit type A.

The moral behind this experiment is that excellent acoutics are essential for such tests; furthermore, the loudspeaker used can have profound effects on the result and should therefore be beyond reproach.

Profiting by this experience a further critical test for loudspeakers was evolved. Since the balance or the microphone position depends very much on the loudspeaker, it was decided in future always to allow the staff responsible for balancing to try new loudspeakers. When this is done the microphone position corresponding to each loudspeaker will vary, and therefore only those loudspeakers which correspond to approximately the same microphone position can be used with safety. Any loudspeaker which makes the operator place the microphone in an unusual position, for example, very close to the orchestra, is suspect, because obviously corrections are being made for deficiencies in the loudspeaker.

Another important test for all electro-acoustic equipment is to use speech from a non-reverberant room or from outdoors. This becomes a very stringent test when the speaker's voice is known. It is surprising that in spite of its simplicity and value very few manufacturers make use of such a speech test from the open air. A skilled observer can pick out quite easily the regions in which colourations exist, but he will frequently find it difficult to convince other people.

An excellent demonstration can be organised by placing a speaker in the non-reverberant room and recording his voice, replaying it into the non-reverberant room on the loudspeaker under test, recording and replaying again and again. This operation, which is equivalent to dubbing in recording systems, accentuates the peculiarities and makes them very audible. All defects of the system are magnified by this process, but as the loudspeaker is the weakest link in the chain its faults predominate.

The loudspeaker tests described serve to illustrate the difficulties of subjective testing and particularly the need for a reliable source of programme. This is not generally realised by manufacturers, who frequently carry out tests on broadcast programmes or on gramophone records. They are then entirely in the hands of the broadcasting organisation or the recording company, which, for example, may have used a loudspeaker with peculiar properties in arriving at the balance. It is not an exaggeration to say that if such a loudspeaker is employed it is only possible to get correct results by employing a similar loudspeaker for listening.

Similar methods are adopted in the subjective testing of microphones. Here again the best method is to use speech, and also to use the microphone on an orchestral concert in a good studio. It is sometimes more convenient to make comparisons at rehearsals. In this connection a word of caution is necessary, for orchestras frequently do not play well at rehearsals and some of the faults, for example, poor intonation between instruments in the same section, can cause a roughness which may sometimes be mistaken for non-linear distortion.

Recording systems can also be tested by comparing the original material with the recorded and reproduced result. In this case dubbing is a very potent method of showing up deficiencies in the recording system. It is usual to find that after four dubbings the faults in even the best recording system have become noticeable.

Thanks are due to the Chief Engineer of the B.B.C. for permission to publish this paper.

A New Scale for Loudness

by F. Langford-Smith

It is common practice to express the loudness of a sound or noise in terms of the sound pressure level of an equally loud pure tone having a frequency of 1000 c/s — this is the phon scale (Ref. 1). The difficulty in using the phon scale is that the sensation of increase in loudness when listening to a change, e.g., from 10 to 40 phons, is quite different from the sensation of increase in loudness from 70 to 100 phons.

In order to meet this difficulty, the loudness unit was standardized (Ref. 3 and 4), based on the principle that doubling the number of loudness units is equivalent to a sensation of twice the loudness. At a later date the loudness unit was replaced by the sone, the sone being defined as the loudness of a 1000 c/s tone, having a loudness level 40 db above threshold, i.e., 40 phons. The same level may also be expressed as 1000 loudness units, hence 1 sone = 1000 loudness units. The loudness level in phons may be measured objectively with a suitable instrument to a reasonably high degree of accuracy — it does not depend in any way on the impressions of a listener. On the other hand, the subjective effect of loudness on the listener cannot be measured with any instrument and is difficult to determine with any great accuracy.

Various attempts have been made to determine the true relationship between subjective loudness and the objective loudness level in phons. The American standard (Ref. 3) is based on the work of Fletcher and Munson (Ref. 2) and is the only widely accepted standard at the present time. However, many investigators have obtained results indicating that the American standard is considerably in error. Two of the most recent of these investigators are Robinson (Ref. 6) in England and Garner (Ref. 7) in U.S.A. It has been demonstrated that there is a considerable difference between loudness doubling and loudness halving, the difference being greatest at low levels, and Robinson gives separate curves for the two conditions (Ref. 6, Figs. 3 and 4).

Garner (Ref. 7) has suggested the use of the name lambda (λ) unit to distinguish it from the sone, since the discrepancy is so wide. Garner's curve is shown with a dashed line in Fig. 1 and may be compared directly with the American Standard.

The difference between the American Standard system and the new scale by Garner is illustrated for the case of doubling the loudness at three representative levels in the table below:

Original loudness level (phons)	Increase in loudness level in phons to give double loudness	
	Standard system	Garner
30	7	19
50	11	23
70	8	23

Robinson's results are indicated in a similar way, to permit direct comparisons:

Original loudness level (phons)	Change in loudness level in phons	
	Loudness doubling	Loudness halving
30	17.5	8
50	17.5	11.5
70	13	12.5

The differences between these results are so serious that the American Standard system must be looked upon with grave suspicion and only used with caution. On the other hand, there is at present no generally accepted alternative. Garner's work was specifically to eliminate, as far as possible, the errors caused in earlier investigations from false assumptions and interfering circumstances, and must be given considerable weight. Further developments are anticipated.

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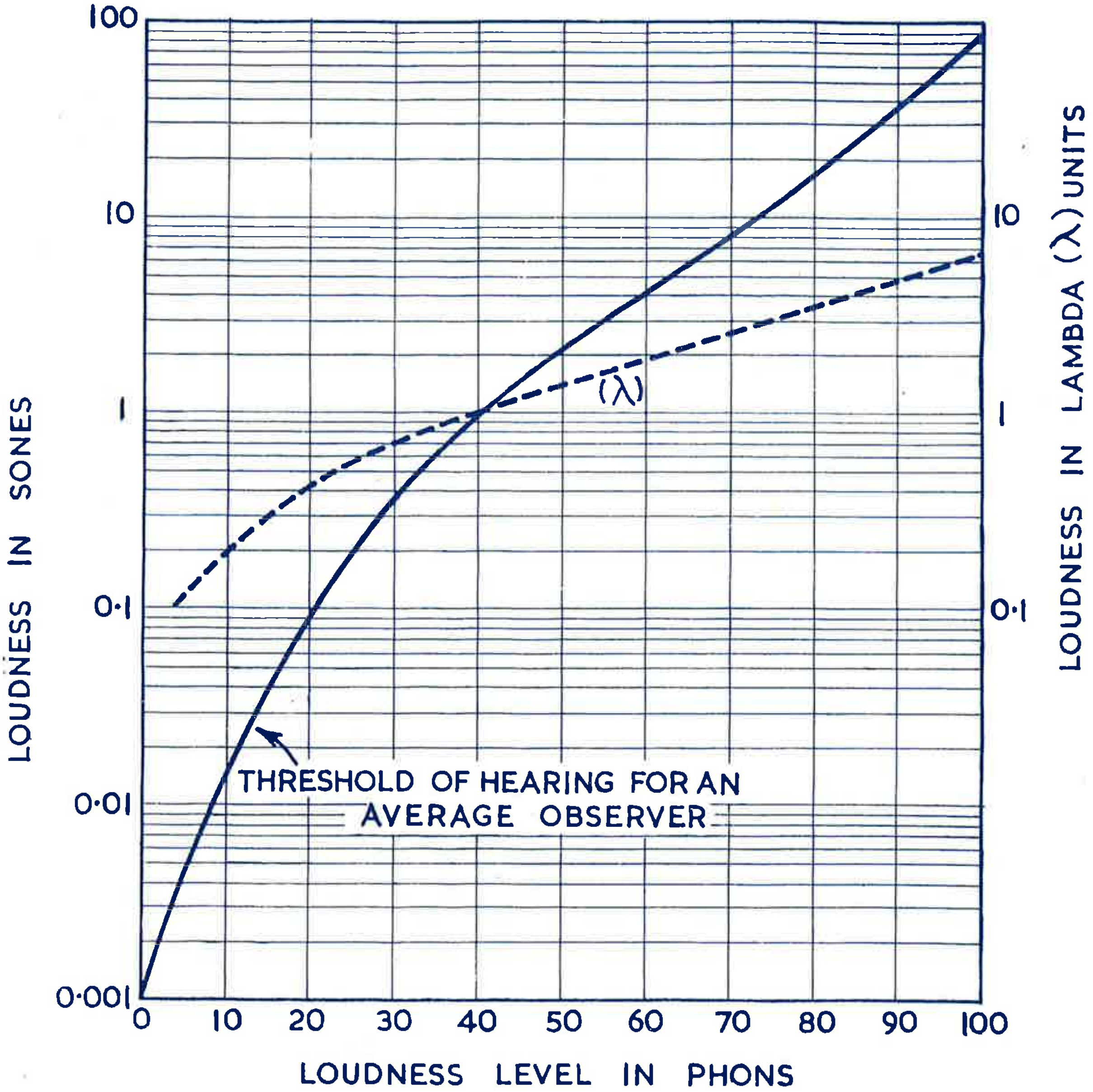

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Fig. 1. Solid curve. Relation between loudness in sones and loudness level in phons (American Standard Z24.2-1942, Ref. 4). Dashed curve: Relation between loudness in lambda units and loudness level in phons (after Garner, Ref. 7).

THE F.M. RECEIVER

by Kenneth Fowler

Comparison with the A.M. receiver

In Section I, the fundamental differences between F.M. and A.M. were explained so that we can now consider what is required of an F.M. receiver.

Figure 13 compares an F.M. receiver and an A.M. receiver in block diagram form. With the exception of the waveshapes at the stages, the only apparent differences in the A.M. and F.M. receivers are the blocks representing the limiter and discriminator stage of the F.M. receiver. However, in addition, there are other modifications in circuit characteristics which make an F.M. receiver considerably different. The principal differences are as follows:

(1) The F.M. receiver uses a different type of 2nd detector, known as a discriminator, which responds to frequency-modulated signals and when perfectly balanced eliminates amplitude-modulated signals.

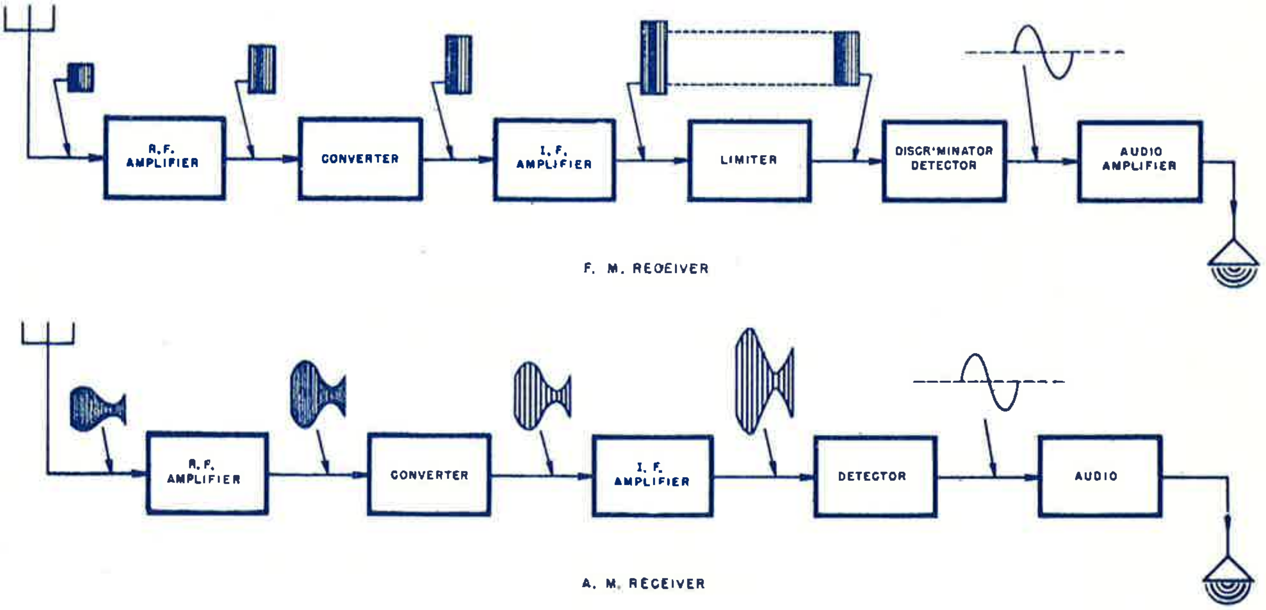

Fig. 13. Block diagrams of F.M. and A.M. receivers.

(2) The use of one or two modified stages in the intermediate frequency amplifier just preceding the discriminator (2nd detector) which act as voltage limiting stages and are known as limiters.

(3) The over-all gain of the receiver, from antenna to the input of the 2nd detector (discriminator), is usually greater than in an ordinary A.M. receiver.

(4) The intermediate frequency is considerably higher than that used in A.M. receivers, being in most cases above 4 mc.

(5) Since the bandwidth requirements of an F.M. signal are much wider than that used in A.M., the frequency response of the intermediate frequency amplifier must be wide enough to pass the total range through which the F.M. signal deviates.

(6) The amount of gain in the radio frequency stage is relatively small, due to the comparatively high frequencies at which it operates.

Requirements of the R-F circuits

The antenna and r-f circuits used in F.M. receivers perform the same general functions as those used in the ordinary A.M. receiver, i.e., to improve selectivity, increase the sensitivity, and reject the image frequency.

One of the important requirements of the r-f circuit for an F.M. receiver is that its frequency response be broad enough so that uniform amplification of all deviation frequencies due to modulation is obtained, as shown in Figure 14. At the r-f frequencies used for F.M. broadcasting, this requirement presents no problem and it is not necessary to take any special precautions. It will be recalled that the intensity or amplitude of the audio level in F.M. depends upon the amount of frequency deviation of

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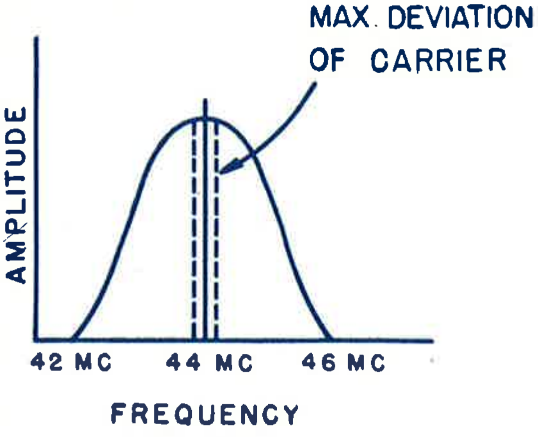

Fig. 14. Typical response of R-F stage in F.M. receiver.

the carrier, and if all the frequencies over which the carrier deviates due to modulation are not amplified an equal amount by the r-f circuits then it is obvious that the variations in the audio level or amplitude in the output of the receiver will not be the same as those transmitted. This results in the distortion of the audio level as transmitted.

A majority of the receivers in use are connected to a dipole antenna by a balanced transmission line system. When this system is used, the antenna input circuit makes use of a low impedance primary winding on the antenna transformer which is balanced to ground by means of a centre-tap on this winding. Another system is to use an unbalanced coaxial or parallel lead line so that one end of the input coil can be connected to r-f ground.

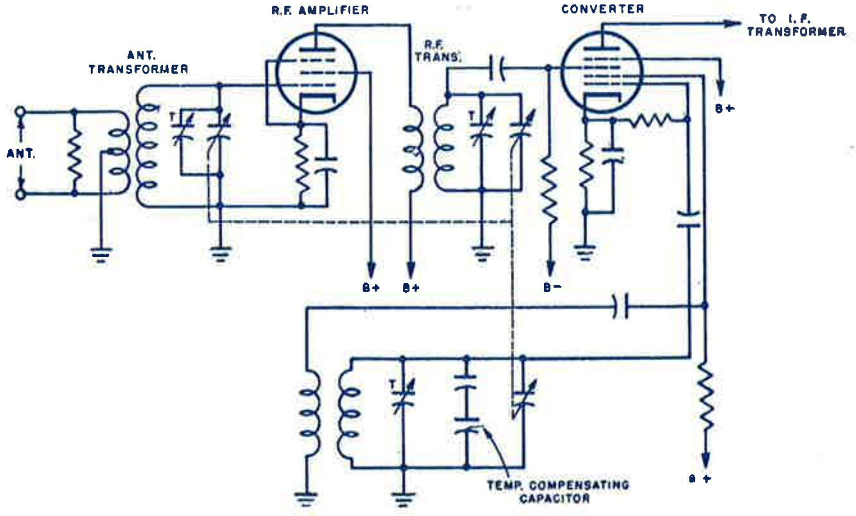

Fig. 15. R-F circuit.

Typical R-F circuits

A typical R-F and Converter Circuit used in an F.M. receiver is shown in Figure 15. Other bands for the reception of A.M. signals may be included, but, for the sake of simplicity, only those circuits that are used when the set is switched to F.M. is shown. Switching from one band to another is done in the conventional manner so there is no need to go into detail regarding it.

As shown, the r-f circuit is very similar to that used in A.M. receivers, except that it is designed to operate at a much higher frequency than the r-f stage in a conventional broadcast receiver. At the high frequency at which the r-f amplifier operates, the selectivity is sufficiently broad enough to pass the required 200 kc pass band. It will be noted that the antenna input is balanced to ground by means of the grounded centre tap on the primary of the antenna transformer to reduce noise pick-up on the balanced transmission line lead-in from the antenna.

The amount of gain that can be obtained from an r-f stage operating at these high frequencies is relatively low due to the losses caused by the low impedance of the tube and stray capacities at these frequencies. Without introducing regeneration, the gain usually does not exceed 10 per stage.

The mixer and oscillator circuits

As shown in Figure 15, the mixer and oscillator circuit, like the r-f circuit, is very similar to those used in the conventional superheterodyne receiver. However, stability of the oscillator circuit is of much greater importance as the operating frequency is raised if a continuous retuning of the oscillator during operation is to be avoided. To keep drift to a minimum, some precautions include the use of low loss insulation on the gang condenser and coil forms, the use of an air condenser for trimming, careful placement of parts and, in addition to this, the use of a temperature compensating capacitor to bring the total drift as near zero as possible. A temperature compensating capacitor drifts in a direction opposite to that of the other components and if properly chosen will keep drift to a minimum. As shown in Figure 15, a temperature compensating condenser is connected in the oscillator tuned circuit to compensate for drift as the receiver warms up.

The steady signal from the oscillator is mixed with the varying r-f signal in the mixer portion to produce an i-f signal which varies above and below the mean i-f frequency.

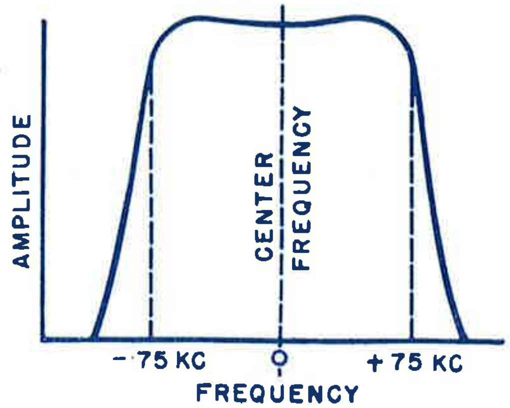

Fig. 16. Ideal i-f selectivity curve.

Requirements of the i-f circuits

The intermediate frequency amplifier of an F.M. receiver must have a broader response than that used in A.M. receivers and it should have a uniform response over the entire range of frequencies contained in the varying transmitted signal. In other words, the ideal frequency response of the i-f amplifier should be flat topped for about 150 kc and then drop off rapidly beyond this range as shown in Figure 16. However, such an ideal response is not practicable and is compensated for as shown later when we take up the action of the limiter. Due to the comparatively high i-f used in F.M. receivers, the stage gain is rather low and also there is a tendency toward regeneration unless particular care is taken in the design of the i-f circuits.

The i-f amplifier in F.M. receivers must also be able to reject interfering signals just as in A.M. receivers and therefore its response should not be any broader than is necessary to pass the band of frequencies necessary for satisfactory reception of F.M. signals.

It has been general practice to employ three i-f stages, including the limiter. This is a greater number than is used in A.M. receivers and is due to the fact that a high signal level is required for efficient operation of the limiter and also because of the lower gain per stage at the higher i-f frequencies used in F.M. receivers.

The centre frequency of the i-f should have as low a value as possible, since the voltage gain per stage of an i-f amplifier is reduced as the frequency is raised. It should not, however, be reduced to the point where image difficulties are encountered. There are many considerations involved in the selection of a suitable i-f. For the new high frequency F.M. band of 88 to 106 megacycles, an intermediate frequency of 10.7 megacycles is used. However, quite a few F.M. receivers were earlier manufactured with an i-f of 2.1 mc and 4.3 mc.

Typical i-f circuits

Figure 17 shows the i-f stages used in a typical FM-AM receiver. The circuit shown is for a combination F.M. and A.M. receiver using the same tubes as far as possible for both services. It will be noted that the i-f transformers for both F.M. and A.M. are connected in series. This is possible since the i-f transformers are parallel resonant circuits and the resonant frequency of the i-f transformers used for the F.M. channel is much greater than that used in the i-f transformers for the A.M. channel (455 kc). The i-f transformer resonated at 455 kc will offer no interference to currents at the F.M. i-f frequency, and likewise the i-f transformer resonated at the F.M. i-f frequency offers no interference to currents at 455 kc. This is true because parallel resonant circuits are practically short circuits to any other frequency than their resonant frequency.

As shown when the band switch is in the position for F.M. reception, the primary of the 1st F.M. i-f transformer T_4 is connected to the plate of the converter and an F.M. i-f voltage appears across the secondary of this transformer and is amplified by the 1st i-f tube. This i-f signal is further amplified and appears across the secondary of the 3rd F.M. i-f transformer T_{11} and is applied to the grid of the limiter.

When the bandswitch is in the position for A.M. reception, the primary of T_4 is disconnected from the plate and instead the primary of the first A.M. i-f transformer T_6 is connected to it. The A.M. i-f voltage of 455 kc passes through T_6, T_7 and T_8 and is demodulated by the diode.

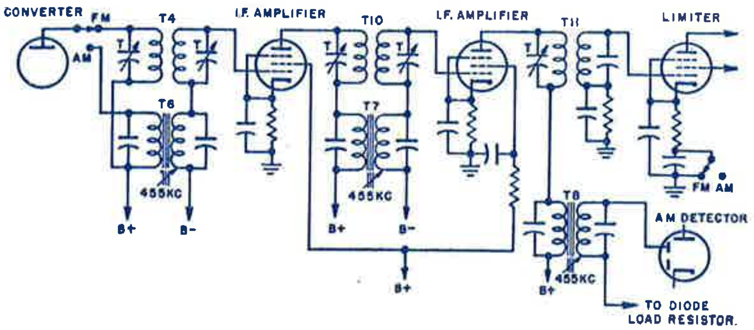

Fig. 17. I-F circuit.

The limiter stage

Thus far, the various circuits that we have covered in the F.M. receiver have been pretty much the same as those used in the ordinary A.M. receiver except for the special requirements mentioned above. However, the limiter stage is considerably different from anything used in conventional receivers, although it is a part of the F.M. i-f system.

The limiter is part of the final i-f amplifier stage and its main function is to remove all amplitude variations that might be present in the i-f signal up to the input of the limiter and to feed a signal to the discriminator circuit that is constant in amplitude. In other words, it limits or shaves off any peaks that might be present on the i-f signal due to noise. This is necessary since any amplitude variations in the signal reaching the discriminator would result in distortion and noise in the output of the receiver.

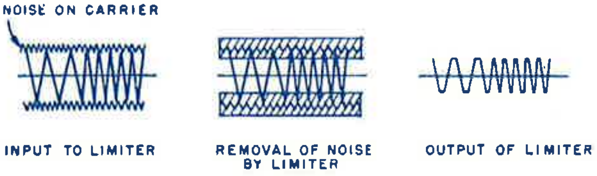

Fig. 18. Limiter operation.

A good example of this action is shown in Figure 18, which compares the limiter to a gate which removes all amplitude variations in the signal above a certain level as determined by the gate and passes a signal that is constant in amplitude.

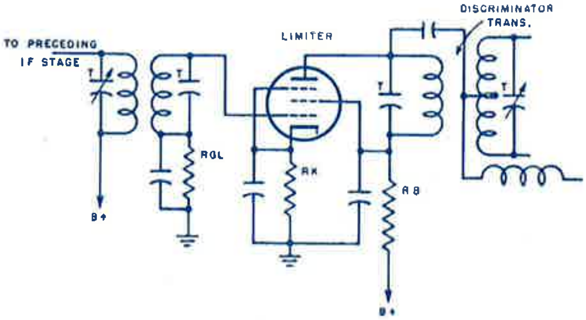

Fig. 19. Limiter stage.

Figure 19 shows a typical limiter stage and is essentially an amplifier which is very easily overloaded. A sharp cut-off tube is used and the plate and screen are operated at low voltages by dropping the voltage through R_B. Thus, with a small signal on the grid, it will produce plate current saturation and causes plate-current cut-off when the signal increases negatively. This action is shown by Figure 20, and it will be seen that with the proper choice of bias and maintaining the plate-screen voltages at a low potential, both halves of the plate current cycle will be equal, but the peaks will be limited or flattened out. This action, therefore, removes any variations in the amplitude of the signal applied to the limiter grid which are greater than those required for plate-current cut-off and plate current saturation.

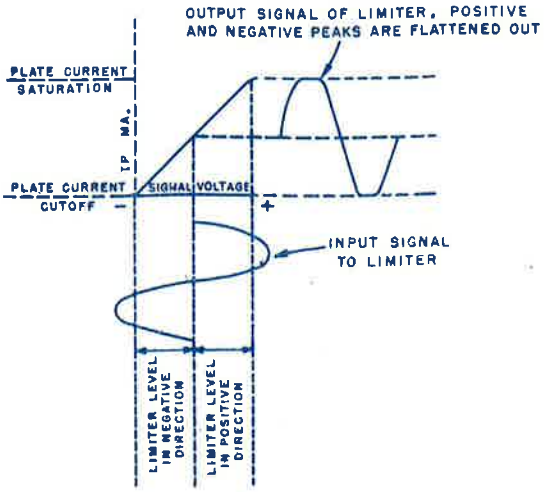

Fig. 20. Output vs. input signal—limiter stage.

At first glance, it would seem that the action of the limiter in flattening the plate current peaks would introduce severe distortion in the output of the receiver, since the plate current variations are not exact reproductions of the grid-voltage variations. This would be true in an A.M. receiver; however, in F.M. the modulation component is contained in the frequency deviations of the signal rather than in amplitude variations of the signal and the flattening of the plate current waveform will not introduce audio distortion. The relative frequencies present in the frequency deviation due to modulation are not affected by the action of the limiter.

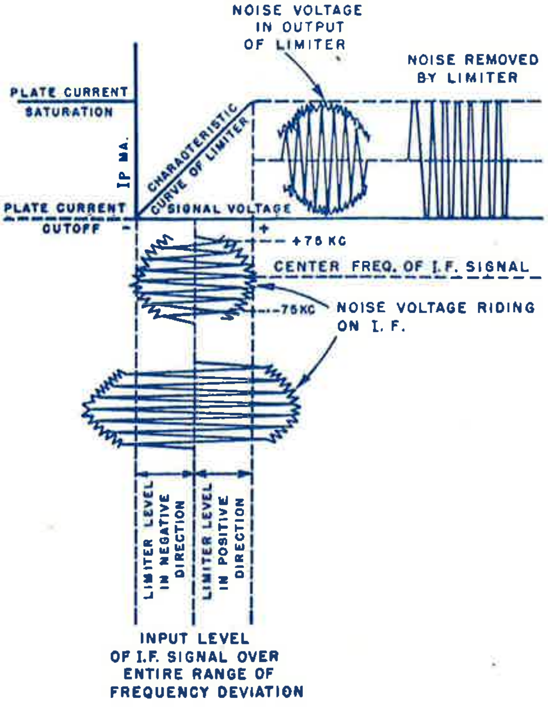

Fig. 21. Limiter operation at varying signal levels.

An important point to bring out in connection with satisfactory limiter action is in the amount of signal voltage that is required at the input to the limiter. If the signal at the limiter grid is too low for satisfactory limiter action, it will cause two undesirable effects. First of these is that any amplitude variations in the signal due to noise will not be clipped off, and the second is that the dynamic range of the signal as transmitted is not duplicated in the receiver.

The first of these undesirable effects can best be explained by referring to Figure 21, which shows signals of varying amplitude at the input to the limiter and the resultant variations in plate current response in the limiter output. As shown, a signal of low amplitude, that is entirely below the level of the limiter action, will be amplified in a linear manner along with whatever noise peaks that might be present. However, if the i-f signal has been amplified enough to give good limiting action, then the output of the limiter will be of constant amplitude and any noise peaks that were present in the input will be eliminated. It is for this reason that it is so desirable to have as much gain as possible in the r-f and i-f stages of an F.M. receiver. It was mentioned in the section on i-f amplifiers that the ideal response curve would have a flattened top for a range of 150 kc and then slope off suddenly with steep sides. However, this ideal is not realized and the response looks more like that shown in Figure 22. Now if this signal were so low that it was below the level of the limiter action, then it would be amplified in the plate circuit in a linear manner and would appear as shown in Figure 23. This means that the various frequencies making up the total frequency deviation would not appear in the discriminator at the same relative amplitude and would cause severe distortion in the output. However, this can be overcome if the i-f signal has been amplified sufficiently so that the voltage level at the input to the limiter of the greatest frequency deviation component is greater than the limiting level. This condition is shown in Figure 24 and the amplitude of the limiter output as shown is constant so that a total band of 150 kc is passed at a constant level and all the frequencies making up the total deviation will be reproduced in their proper relation.

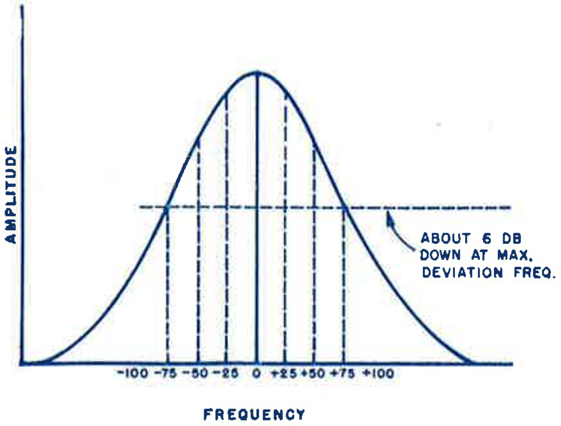

Fig. 22. Typical I-F Selectivity Curve.

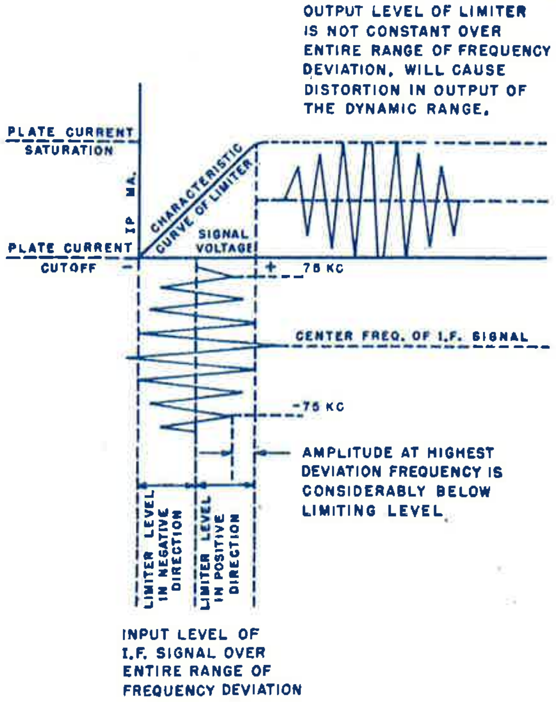

Fig. 23. Limiter action on insufficient signal.

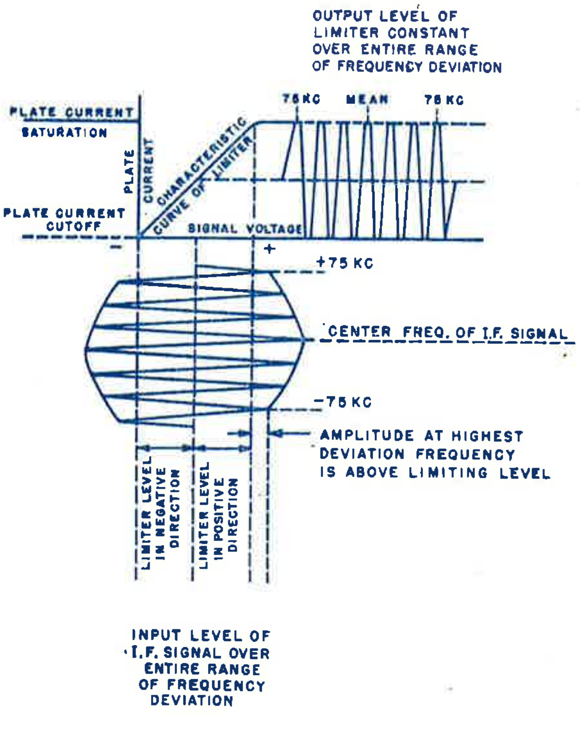

Fig. 24. Limiter action on strong signal.

In summation, the functions of the limiter are: (1) Remove all amplitude variations of the signal due to noise; and (2) compensate for the lack of an ideal flat-topped response of the i-f stages, provided that the i-f signal at the input to the limiter is high enough at all frequencies within the complete frequency deviation range so that limiter action takes place over the entire range of frequency changes from no modulation up to full modulation, and, also, that the limiter be designed to give a constant output over a wide range of input voltages; and (3) reject unwanted signals 2 to 1 or more reduced in amplitude. The limiter is placed in the final i-f stage and its output is fed into the discriminator-2nd-detector stage.

The discriminator stage

The second point at which the F.M. receiver is considerably different from an A.M. receiver is in the method used to demodulate the i-f signal. Since in F.M. the modulating voltage or audio component is contained in frequency deviation of the signal, then some means of converting these frequency changes back to voltage changes that vary in accordance with the audio or modulating voltage must be employed. The circuit used for this purpose is known as a discriminator. Fundamentally, the discriminator performs the same function as the 2nd detector in an ordinary superheterodyne circuit but its action is quite different.

The discriminator is not new in radio receivers. It is used as a means of developing a control voltage in receivers which employ automatic frequency control systems.

There are two well-known circuits in use, namely, the Round (Travis) circuit and the Foster-Seeley circuit. Since the former is the easiest to understand, it will be explained first.

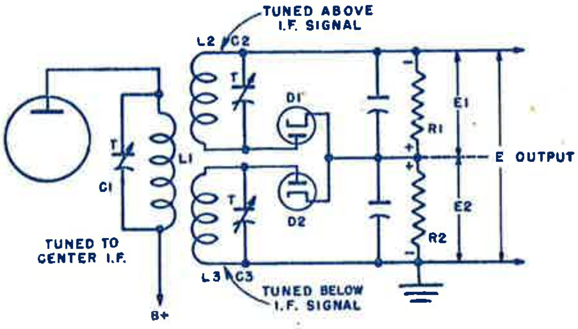

Fig. 25. Discriminator circuit.

As shown in Figure 25, the discriminator transformer consists of a primary L_1C_1, which is tuned to the mean or centre frequency of the i-f signal. There are also two secondary windings, one of which is tuned to a frequency slightly above the centre of the i-f signal and the other is tuned to a frequency slightly below the centre frequency. In Figure 25, the secondary tuned circuit L_2C_2 is resonated slightly above the mean i-f and the secondary tuned circuit L_3C_3 is resonated slightly below the mean i-f. The signal voltage appearing across the secondaries is shown by Figure 26A. It can be seen that when the i-f passes through its centre frequency, equal voltages will appear across each secondary. However, if the frequency deviates above the centre of the i-f, then a greater voltage will appear across the secondary tuned circuit L_2C_2 than that across L_3C_3. This is so because L_2C_2 is tuned to a higher frequency than L_3C_3. Likewise, if the frequency deviates below the centre of the i-f, then a greater voltage will appear across L_3C_3 than across L_2C_2.

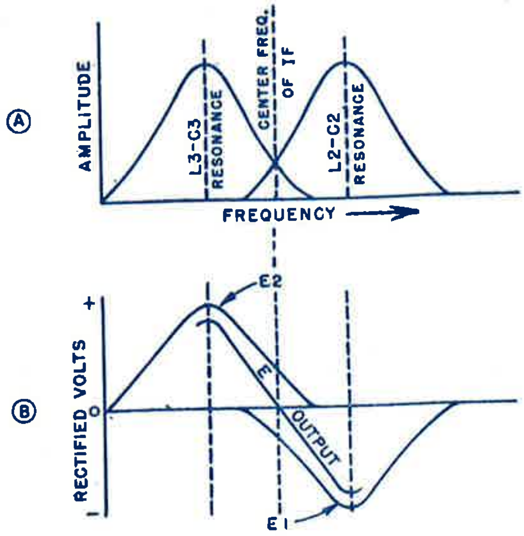

Fig. 26. Discriminator resonance vs. output curve.

This means that when the i-f is at its centre frequency, each diode will conduct equally and will develop equal but opposite voltages across each diode load resistor. Thus when diode D_1 conducts the current will flow from its cathode to the diode load resistor R_1, through R_1, and back to the other end of L_2, thus making the cathode end of R_1 positive with respect to its other end by an amount equal to the IR drop across the resistor. When diode D_2 conducts, the current flows from its cathode through diode load resistor R_2 and back to the other end of L_3, making the cathode end of R_2 positive with respect to the end that is connected to ground. The IR drop across R_2 will be the same as that across R_1, but of opposite polarity, since each load resistor has the same value and equal voltages appear across each secondary when the i-f is at its centre frequency. In Figure 25, the voltages developed across the diode load resistors are represented by E_1 and E_2 and have opposite polarity as shown. The total voltage appearing across both diode resistors as represented by "E output" will be zero since the voltage across each resistor is equal and opposite and therefore cancels each other.

Now if the i-f deviates above its centre frequency, a greater voltage will appear across L_2C_2 than that across L_3C_3, and therefore diode D_1 will conduct more than diode D_2. This will develop a greater voltage across R_1 than that across R_2 and the polarity of the voltage across each resistor will still be opposite as shown in the figure. However, the total voltage "E output" appearing across both diode resistors will no longer be zero, but will have a value and polarity equal to the algebraic sum of the voltage appearing across each diode load resistor. Thus, if in the above case the voltage developed across R_1 is 5 volts, it can be expressed as —5 volts with respect to ground. Also, if the voltage developed across R_2 is 3 volts, it can be expressed as + 3 volts with respect to ground. This is so because, regardless of the amount of voltage developed across each resistor, the polarity across each resistor with respect to ground will always be the same as shown in Figure 25. If we add the voltage across each resistor algebraically, then in the above example the total voltage across both resistors will be the elgebraic sum of $-5 + 3$, or -2, volts with respect to ground.

Now when the i-f deviates below its centre frequency the same amount as it did above its centre frequency in the previous example, then a greater voltage will appear across L_3C_3 than that across L_2C_2 just the opposite from before. This time the voltage appearing across R_1 will only be 3 volts instead of 5 as before, and that appearing across R_2 will be 5 volts. Therefore when we add algebraically the voltage across R_1 (which is -3 volts) with respect to ground and the voltage across R_B (which is $+5$ volts) with respect to ground, we get a total voltage of $+2$ volts from "E output", which is just the opposite from before.

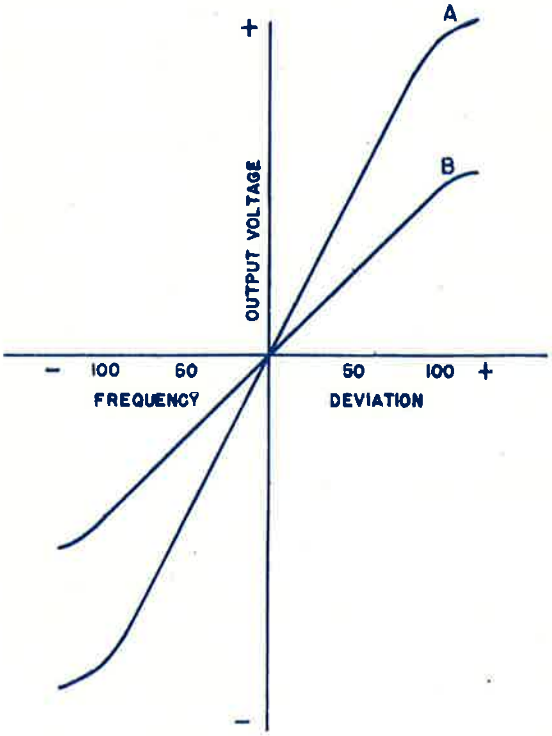

Fig. 27. Characteristic Curve for an F.M. Discriminator.

This action is shown by Fig. 26B, which shows the rectified voltage developed across each diode load resistor with respect to deviations in frequency and also the overall output of the discriminator as shown by curve marked "E output".

It can be seen from the foregoing that the total voltage output of the discriminator will vary in a plus and minus direction, depending on the amount that the i-f signal deviates on either side of its mean or centre frequency. The greater the signal deviates the greater will be the amount of voltage developed. It is therefore apparent that the voltage output of the discriminator will vary in exact accordance with the audio voltage that modulates the transmitted carrier. There is a very important point to bring out here and it is that the output voltage of a discriminator can also vary directly with any change of input voltage. This fact was not apparent in the above discussion on the action of the discriminator since we stressed the effect of frequency deviations of the input signal on the discriminator output, with the input level held at a constant amplitude through the action of the limiter.

Figure 27 shows the typical characteristic curves for an F.M. discriminator at two different values of input voltage. As shown, the output voltage of the discriminator is greater for a higher input level as represented by curve "A" than for a lower input level as represented by curve "B".

This means then that the output voltage of a discriminator can vary, not only in accordance with deviations in frequency of the input signal but also in accordance with variations in the level of the input signal. However, under normal conditions the input to the discriminator will be held at a constant level by the action of the limiter and therefore the discriminator output will vary only in accordance with deviations in frequency of the input signal which is the desired condition.

If for some reason, such as for insufficient signal voltage to the input of the limiter, or due to some fault of the limiter, the input level to the discriminator is not constant, then the output of the discriminator will no longer reproduce the dynamic range of the signal as transmitted since it will contain voltages that are due to amplitude variations as well as frequency variations and severe distortion is likely to result. Also, if any noise is present in the signal, it too will be reproduced.

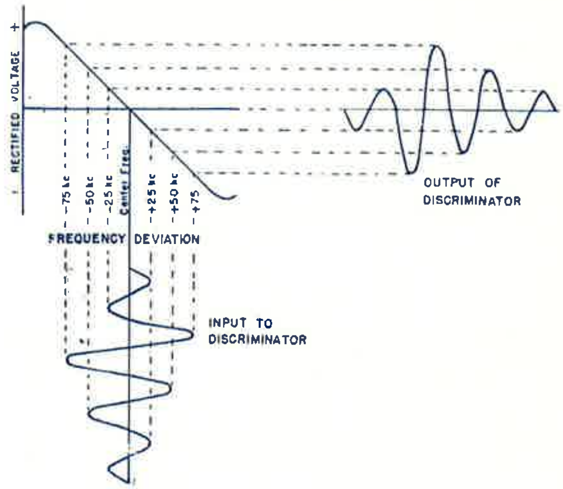

Fig. 28. Conversion of Frequency Changes into Audio Voltage.

Figure 28 shows how frequency deviations of the i-f signal are converted into corresponding audio signals by means of the linear portion of the discriminator characteristic. It is thus important that the discriminator output be linear over a wide range so that the frequency variations will take place over the straight portion of the discriminator characteristic. If the frequency variations take place over a non-linear portion of the discriminator characteristic, distortion will take place just as in an audio amplifier when the signal voltage swings past the linear portion of the amplifier characteristic. It is for this reason that the discriminator characteristic is made linear for considerably more than the maximum frequency deviation of ± 75 kc. This also alleviates the effects of drift and provides less critical tuning.

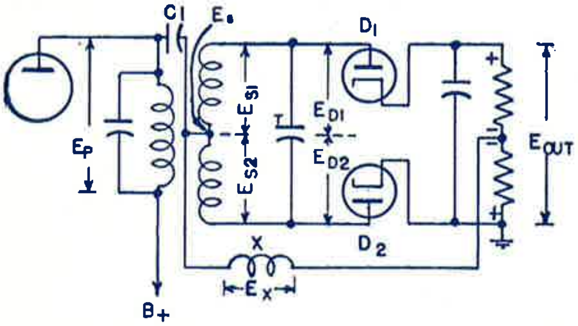

Fig. 29. Foster-Seeley Discriminator Circuit.

The discriminator circuit just described is not as popular as the Foster-Seeley type discriminator shown in Figure 29, but everything that was said concerning the development of voltage across the diode load resistors holds true. The only difference is in the way the discriminator transformer responds to deviations in the i-f signal.

As shown, the discriminator transformer has only one secondary and has a centre tap which is coupled to the plate side of the primary through a capacitor. The primary and secondary are both tuned to the mean or centre frequency of the i-f signal. The action is as follows:

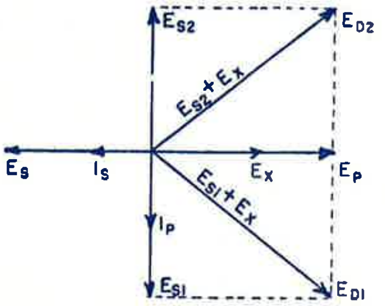

Fig. 30. Vector Diagram at Resonance.

It is assumed that the i-f signal is at its mean or centre frequency and the discriminator action will be shown by means of vector diagrams. Referring to Figure 30, the voltage E_p appearing across the primary is drawn as a horizontal vector to the right of the origin. Since the current through a coil lags the voltage by 90°, the vector representing the current in the primary I_p is drawn at right-angles to E_p and 90° lagging. It is assumed that the vectors are moving in a counterclockwise direction. This current I_p in the primary produces a flux which in turn causes a voltage to be built up in the secondary winding. This voltage induced in the secondary winding will be a maximum when the flux is changing most rapidly and this is the point at which the current wave crosses the zero axis, as shown in Figure 31. The voltage therefore induced in the secondary winding will be 180° out of phase with the primary voltage. This vector E_s is therefore drawn on the horizontal but to the left of E_p.

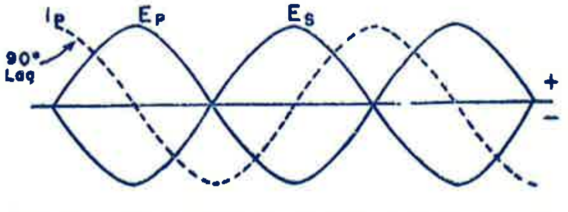

Fig. 31. Phase Diagram of Discriminator E_p, E_s, and I_p.

At resonance. the capacitive and inductive reactances of any tuned circuit are equal and opposite and, therefore, cancel each other, making the tuned circuit resistive instead of reactive. This means that the circulating current in the secondary tank circuit will be in phase with the induced secondary voltage, therefore the vector for the secondary tank current I_s is drawn parallel with the vector E_s. Now this circulating current I_s in the secondary tank circuit develops another voltage in the secondary circuit which is considerably greater than the voltage induced in the secondary winding and its magnitude is equal to the product of the circulating current and the inductive reactance of the secondary winding, i.e., I X. The phase relation between E_p, E_s and I_p is shown in Figure 31.

If we now consider the top half of the secondary winding separately, we find that the circulating current will cause a voltage drop across it that will lead the circulating current by 90° and is marked as E_{s1}. This is so because the voltage across an inductance always leads the current by 90° provided there is negligible resistance in the coil. Now we know that the voltage across the bottom half of the secondary must be 180° out of phase with that appearing across the top half and therefore is marked as vector E_{s2}.

Now that we have established the phase relationship between the voltages and currents existing in the discriminator transformer, we can consider the effect of the voltage coupled from the primary to the secondary of the discriminator transformer by means of capacitor C_1.

Since coil X is connected to the plate side of the primary, then the voltage E_X across this coil will be in phase with E_p and may be taken as being in series with either half of the secondary voltage. This can readily be seen by tracing the circuit from the diode plate through one-half of the secondary, through coil X, through the cathode resistor to the cathode, and back to the plate. The path will be the same regardless of which half of the secondary is used. Therefore, considering the top half of the discriminator secondary, the resultant voltage of E_X and E_{S1} when added vectorially will lag E_X by 45° since there is a 90° phase angle between the two. This is shown as the vector E_{D1} and is the voltage that will cause current to flow through the upper diode, D1, circuit. The resultant voltage of E_X and E_{S2} when added vectorially will lead E_X by 450°, as shown by vector E_{D2}, and is the voltage that causes current to flow in the lower diode, D2, circuit. Since E_{D1} and E_{D2} are equal, as shown by the vector diagram, then the total voltage across the diode load will be zero, which is the desired condition when the i-f signal is at its mean or centre frequency.

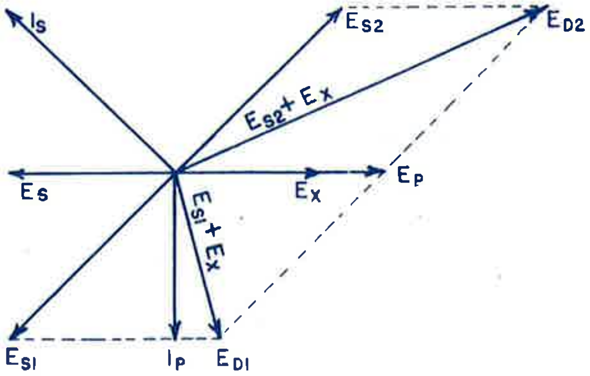

Fig. 32. Vector Diagram above Resonance.

Now let us consider what happens when the i-f signal is above the mean frequency. The phase relationship of E_p, I_p and the induced secondary voltage of E_s will remain the same as before, but the phase relationship of the other voltages will change since a condition of resonance no longer exists in the secondary circuit. The impedance of the secondary will no longer act as pure resistance but will be reactive and, since the i-f signal is above the resonant frequency, the tuned tank circuit will be inductive and the circulating current will lag the induced voltage E_s by an amount depending on the amount that the i-f signal is above resonance. For this case we can assume that the frequency is above the centre or mean frequency by an amount that will cause the current to lag the induced voltage by 45° and is shown by the vector I_s in Figure 32 (above resonance). This current flowing through the upper half of the secondary will cause a voltage drop I_sX_L across it, and this voltage will lead the current by 90° as shown by vector E_{S1}. We know that the voltage appearing across the lower half of the secondary must be 180° out of phase with that across the upper half and is plotted as vector E_{S2}. Considering E_{S1} and the voltage across the coil E_X vectorially, we obtain the vector E_{D1} which represents the voltage applied to diode plate D1. As shown by vector E_{D1}, this voltage is less than either. E_{S1} or E_X due to their phase relationship. Now considering E_{S2} and the voltage across the coil E_X vectorially, we obtain the vector E_{D2} which represents the voltage applied to diode plate D2. This vector is greater than that representing E_{D1}, and, therefore, a greater voltage will appear at diode plate D2 than at diode plate D1. This difference in diode voltage will cause a net voltage to appear across the diode load which will be negative with respect to ground. The greater the frequency deviation above the mean or centre frequency, the greater will be the negative voltage developed across the diode load which is the desired condition when the i-f signal deviates above its mean or centre frequency. The output voltage of the discriminator with relationship to frequency deviation is shown by Figure 34. Again, reversing conditions so that i-f signal is below its centre frequency by the same amount that it was above, we find that the phase relationship of E_p, I_p, and the induced secondary voltage E_s still remain the same as shown by Figure 33. Since the i-f signal is below its centre frequency, the impedance of the secondary-tuned circuit will be capacitive and the circulating current will now lead the induced voltage. This condition is shown by the vector marked I_s which is shown leading the induced

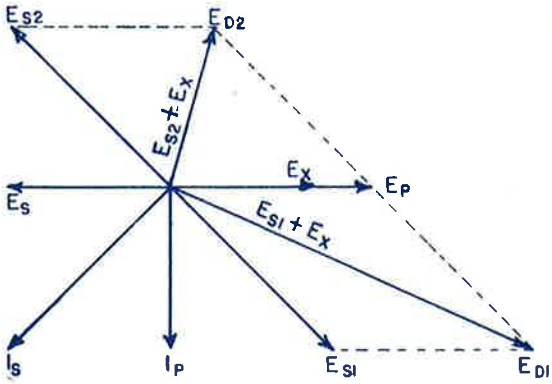

Fig. 33. Vector Diagram below Resonance.

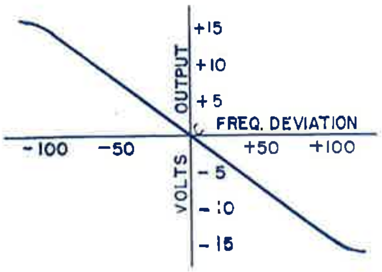

Fig. 34. Discriminator Characteristics.

voltage by 45°. The voltage I_sX_L developed across the upper half of the secondary will lead the circulating current by 90° and is marked as E_{s1} and is so indicated on the vector diagram. The voltage E_{s2}, appearing across the lower half of the secondary, is 180° out of phase with E_{s1} and is so indicated on the vector diagram. Combining E_{s1} and E_X vectorially, we get the vector marked E_{D1} and, as shown, it is greater than either E_{s1} or E_X separately due to their phase relationship. As shown by vector E_{D2}, the voltage appearing across diode D2 will be less than that appearing across diode D1. Therefore, diode D1 will conduct more than D2 and the net voltage developed across the diode will be positive with respect to ground and is just the opposite from the condition obtained when the i-f signal was above its mean frequency. It can therefore be seen that a voltage will appear across the diode load of this type of discriminator which will vary in exact accordance with the frequency deviation of the i-f signal. The greater the frequency deviation on either side of the centre or mean frequency of the i-f, the greater will be the amount of voltage developed across the diode load and therefore the louder will be the sound coming from the speaker of the receiver. Likewise, when there is no modulation on the F.M. carrier, there will be no deviation of the frequency and consequently no audio voltage will be developed across the diode load and no sound will come from the speaker.

The audio system

The audio amplifier system of an F.M. receiver is very similar to that used in any other receiver except for a few differences discussed below.

The audio system in an F.M. receiver is designed to take advantage of the greater frequency range inherent in F.M. Thus, the audio system in an F.M. receiver has a considerably better response than the audio system used in the average A.M. receiver.

The F.M. transmitter amplifies the higher audio frequencies more than the lower audio frequencies and this is known as pre-emphasis. The reason for doing this is to minimize noise, since most of the noise in an F.M. system is concentrated in the higher frequencies. At the receiver, these higher audio frequencies have to be de-emphasised and is done by means of a corrector network which is placed between the output of the discriminator and the audio amplifier. This corrector network merely compensates for the pre-emphasis at the transmitter by attenuating these higher audio frequencies and also whatever noise that might be present at these frequencies. The corrector network restores the relative amplitude of the various frequencies to where it was before pre-emphasis took place at the transmitter.

Special features of F.M. receivers

Thus far, we have covered the fundamental circuits needed for a straight-forward F.M. receiver. However, there are several special features or refinements incorporated in some receivers to give better overall results. Some of these will be described in the following paragraphs.

Cascade Converter Circuit.—The first of these is the use of the double superheterodyne or cascade converter circuit. By using this method of dual conversion, a considerable amount of gain is realised ahead of the i-f amplifier over that which can be obtained by the ordinary method of conversion. This increase in sensitivity that is secured by the use of cascade converter circuit insures adequate receiver sensitivity for proper limiter operation.

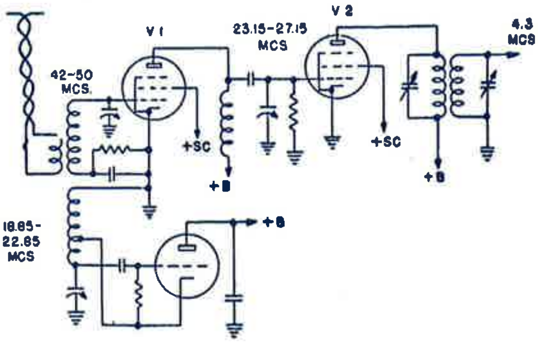

Fig. 35. Cascade Converter Circuit.

The typical cascade converter circuit employs two converter tubes and an oscillator tube, as shown by Figure 35. The tuning condensers for the antenna, r-f, and oscillator are ganged together as usual. The antenna circuit tunes the F.M. band from 42 to 50 megacycles, the r-f circuit tunes from 23.15 to 27.15 megacycles and the oscillator from 18.85 to 22.85 megacycles. The oscillator voltage is inductively coupled to the grid of the first converter tube V1. This produces, by heterodyne action, a signal to which the plate circuit of V1 is tuned. V1 provides a gain of unity for the oscillator frequency and accordingly the oscillator voltage is also applied to the grid of V2.

Tube V2 operates as a second converter and the oscillator signal on the grid of V2 heterodynes with the tuned signal appearing in the plate circuit of V1 which is coupled to the grid of V2 and produces an i-f or 4.3 mc in the output of the second converter, the plate circuit of which is tuned to 4.3 mc.

To illustrate the action, consider an F.M. signal of 42 mc to which the receiver is tuned. The oscillator frequency for this setting of the tuning control is 18.85 mc and it heterodynes in the 1st converter tube, in the usual way, with the 42 mc signal to form a signal of 23.15 mc (the difference between 42 mc and 18.85 mc). This 23.15 mc signal appears on the grid of V2 along with the oscillator frequency at 18.85 mc, and this 23.15 mc signal in turn beats with the oscillator signal of 18.85 mc and produces a 4.3 mc signal in the plate circuit of the second converter tube. Since the primary of the 1st i-f transformer (which is tuned to 4.3 mc) is connected to the plate of the second converter, we have an i-f of 4.3 mc which will be amplified by the i-f system.

Cascade Limiters.—Another refinement is in the use of cascade limiters instead of just one limiter, which provides more satisfactory noise limiting. With the conventional single tube limiter of the grid bias type, the time constant of the grid resistor condenser combination is too slow to properly follow short duration impulse noises, such as result from automobile ignition systems, if the time constant is of such value as to provide for good balance of the limiter over a wide range of input levels at the limiter grid. Therefore, a compromise is made between a good balance and the reduction of impulse noise.

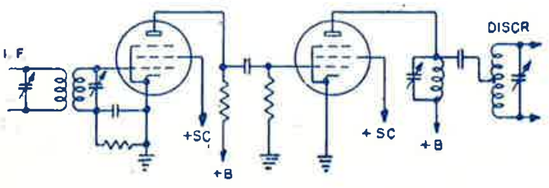

Fig. 36. Cascade Limiter Circuit.

However, the deficiencies described above can be essentially removed by the use of a cascade limiter. Figure 36 shows a typical cascade limiter circuit. The circuit consists of two resistance-coupled tubes in series. Each limiter operates at zero initial bias and low plate voltage. Both grid circuits are designed for self-biasing and the use of capacity-resistance networks provide enough time delay to retain the grid bias between signal peaks. The action of the limiter is such that as soon as a signal is applied to the grid of the tube, the grid draws current. This grid current charges up the capacitor across the grid resistor and at the same time establishes a bias through current drain in the resistor. Between positive signal peaks the capacitor discharges through the resistor maintaining the grid bias. The circuit is so designed that negative signal peaks are all beyond plate current cut-off and positive signal peaks are cut-off by plate current saturation. The time constant of the 1st limiter capacity-resistor network is low enough to limit peaks due to impulse noise satisfactorily. This arrangement leaves the 2nd limiter with the very much simplified task of reducing the remaining noise and provides for good balance of the limiter over a wide range of input levels. In addition, the effectiveness of the receiver on weak signals is improved since there is additional gain provided by the use of two limiter stages.

Squelch Circuit.—As pointed out previously, the limited circuit of an F.M. receiver will only remove noise when an F.M. signal of sufficient strength is present at the input of the limiter to provide good limiting action.

Since the noise limiter circuits only operate when an F.M. carrier is present, noise between stations will ride through with undiminished amplitude. In order to remove this undesirable effect, some F.M. receivers make use of an F.M. station silencer circuit or simply a squelch circuit which operates on the amplitude-modulated noise signals present between stations to produce squelch or quieting of the audio amplifier.

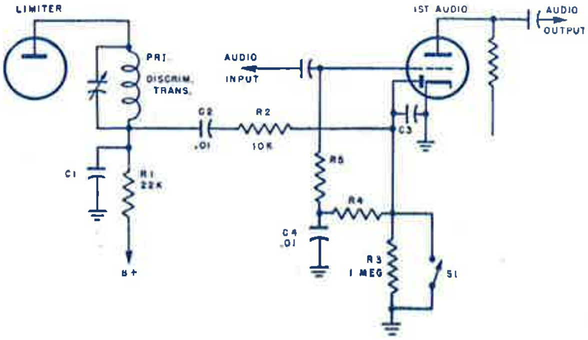

Fig. 37. Squelch Circuit.

A breakdown of a typical squelch circuit used in several F.M. receivers is shown in Figure 37. The noise signal appears in the limiter plate circuit and develops a voltage across R1, which is in series with the primary of the discriminator transformer. This voltage is applied to one diode through C2 and R2, and is rectified by the diode. The current due to rectification of this noise voltage flows from the diode to the cathode which is grounded, through R3, and back to the diode, thus developing a voltage across R3 with the grounded end positive with respect to the end connected to the diode plate. This rectified voltage is applied to the grid of the 1st audio tube through R4 and R5, making the grid more negative than the cathode by an amount equal to the rectified voltage developed across R3. This rectified voltage is sufficient to completely bias off the tube so that no audio signal is passed.

When an F.M. signal is received that is strong enough to satisfactorily operate the limiter, the noise or amplitude-modulated signal is removed altogether or greatly reduced by the operation of the limiter as previously described. Therefore, the voltage developed across R_2 due to a noise signal is removed or greatly reduced so that little or no current flows in the diode circuit containing R_3, and consequently the cut-off bias previously applied to the grid is removed and the tube amplifies the audio signal and passes it on in the normal manner.

If it is desirable to receive F.M. stations that are too weak to satisfactorily operate the limiter, the squelch voltage can be manually removed by closing switch S1, which simply shorts out R3 and places the grid at ground potential. However, considerable noise is likely to be present when receiving such weak signals.

New RCA Releases

The new **Radiotron–6383** — a very compact, liquid-and-forced-air-cooled uhf power triode — is designed primarily for applications where the transmitter design factors of compactness, light weight, and high power output are prime considerations. It has a maximum plate dissipation of 600 watts, and can be operated with full plate voltage and plate input at frequencies up to 2000 Mc.

The design of the 6383 features an integral cooling jacket with associated inlet and outlet pipes, and a coaxial-electrode structure for use with circuits of the coaxial-cylinder type. The structure provides low-inductance, large-area, rf electrode terminals for insertion into the cylinders, and permits effective isolation of the plate from the cathode. The latter feature makes the 6383 particularly suitable for cathode-drive circuits.

Small in size for its power capability, the 6383 has an over-all length of $4\frac{1}{4}$″, a diameter of $1\frac{3}{4}$, and a weight of 8 ounces.

Radiotron–5AUP24 is a five-inch cathode-ray tube intended primarily for use as the flying-spot scanner in a colour video-signal generator. Colour television signals may be generated by scanning Kodachrome slides or similar transparencies.

This new tube features a metal-backed phosphor which has a spectral-energy emission characteristic with peak in the blue-green region and with sufficient range to provide usable energy over the visible spectrum required for generating colour signals from colour transparencies. Because of the extremely short persistence of the phosphor, very little equalization is needed to minimize blurring or trailing in the reproduced picture, and hence good signal-to-noise ratio is obtained.

The tube face is made of a special, non-darkening glass, and has an optical quality which will not limit the performance of a high-quality objective lens needed to provide maximum resolution.

Other features of the 5AUP24 include a high-resolution gun of the electrostatic-focus type; a 40° deflection angle to minimize deflection defocusing and provide high corner resolution; an external conductive coating on the neck which, when grounded, prevents corona between yoke and neck; a built-in capacitance to serve as a supplementary filter capacitor; and an external insulating coating on the bulb cone to minimize sparking over the bulb under conditions of high humidity. The maximum ultor rating (design centre) for the 5AUP24 is 27,000 volts.

corrected.

Correction August issue page 92 right hand column, second line from bottom: Read 100 instead of 1000.

Editor .. .. Ian C. Hansen

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