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FACTORS RELATING TO FAITHFUL REPRODUCTION

BY

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The subject of phonograph or radio reproduction is necessarily a very broad one. For this reason the following paper concerns itself only with aural compensation and volume expansion as aids in obtaining the desired result. Demonstrations will be given of various factors involved and a very useful piece of laboratory equipment in connection with this work will be described and demonstrated.

Before discussing aural compensation as applied to an electric phonograph or radio receiver, it would probably be well to digress a moment and consider the general subject of sound and hearing.

A large amount of research work has been carried on by different experimenters in connection with sound at different levels and its effect upon the human ear. General reference to books and papers on the subject gives one the impression that although some discrepancies exist between results obtained by different experimenters, there is, in the majority of cases, quite close agreement with the work of Dr. Harvey Fletcher of the Bell Laboratories. For this reason we have, in our work on aural compensation, used data presented by Dr. Fletcher on his paper "Loudness, Its Definition, Measurement and Calculation"[1].

Listening carefully to the radio set of a few years ago, we have all undoubtedly noticed that a reduction in volume from a rather loud level to one which could be used in the average apartment without causing annoyance to ones neighbors was accompanied by a change in musical balance. At the louder level the low frequency instruments such as the bass viol, bass drum and tuba were in proper balance with the middle and high frequency instruments. At the lower level, however, the low frequency instruments were no longer in balance and the music had a harsh or tinny characteristic. This effect could also be obtained at an outdoor orchestral performance or band concert if one were to change his position from one close to the instruments to a position a hundred feet removed from them. The high frequency instruments will decrease in volume much less rapidly than the tuba and other low frequency instruments. The reasons for this phenomenon are very clearly given in Dr. Fletcher's paper.

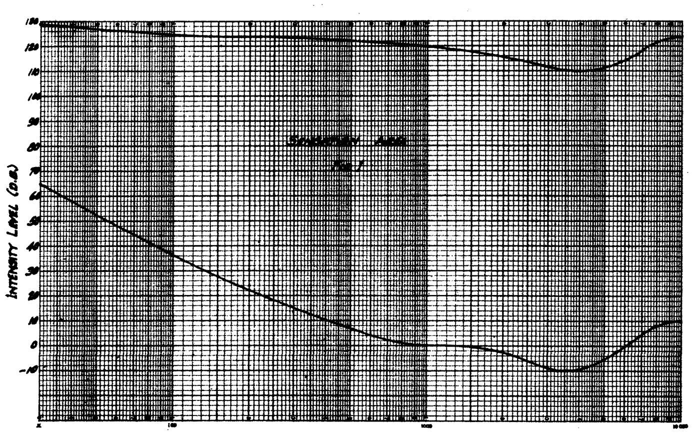

Figure #1 illustrates the frequency response and intensity range of the average human ear from the threshold of hearing to the threshold of feeling. These two thresholds were determined at different frequencies by means of pure tones applied to head phones worn by the observers. Many different readings were taken at various fixed frequencies and for many different observers. From these, it was possible to plot the lower curve which shows the threshold of hearing for the average individual. It will be noted that the curve is plotted with 1,000 cycles as the 0 d.b. level. For reference purposes the input level to the ear canal at this frequency was found to be 10-16 watts per square centimeter. It is easily seen by reference to the curve that for a frequency of 100 cycles, the same intensity level in the ear would require an input 35 d.b. higher than for 1,000 cycles. Conversely, the input in the 3000 - 4000 cycle range would need to be 10 d.b. less than that at 1000 cycles to produce equivalent effect. The upper curve was determined in much the same manner with the exception that the limiting factor became the point at which the observer felt actual pain. This curve, which is called the threshold of feeling, represents the highest level the average individual can withstand for

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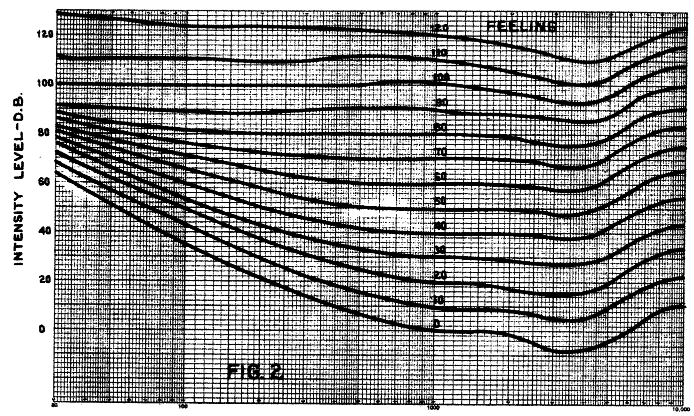
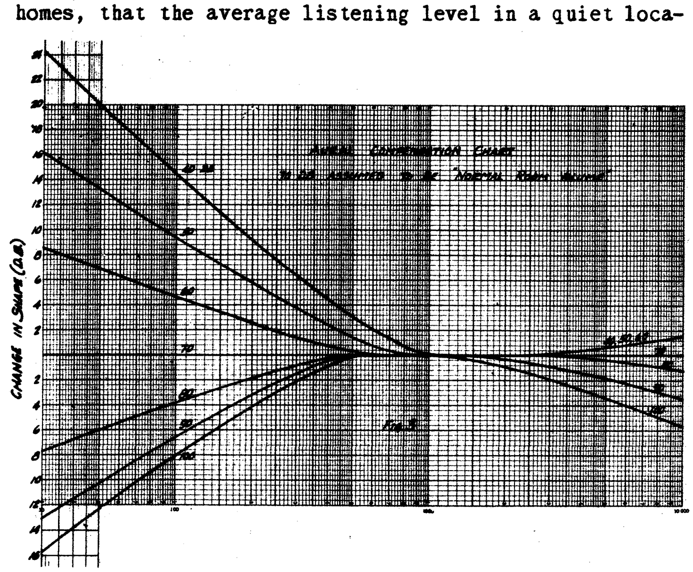
any length of time without actually suffering pain as a result. It will be noticed in this case the input at 100 cycles need be only 5 d.b. higher than that at 1000 cycles to produce the same result. In other words at the threshold of feeling the curve of the average ear is essentially flat.

Between the two thresholds, it then became possible to plot a family of curves representing the average characteristic at different levels. For convenience these curves were determined in 10 d.b. steps at 1000 cycles beginning with the threshold of hearing and carrying up to the threshold of feeling. Figure #2 shows this family of



curves. An observation of these curves indicates immediately that the range between 500 and 5000 cycles is practically flat for almost any level but that for frequencies below 500 cycles, the input to the ear must be increased over the 1000 cycle level to maintain proper balance. These curves then show us that some form of low frequency compensation is necessary if we are to maintain proper balance over very wide changes in volume. It is, of course understood that the volume changes referred to are the changes in average level from one value to another rather than the normal variations in dynamic level which are automatically taken care of by the conductor of the orchestra.

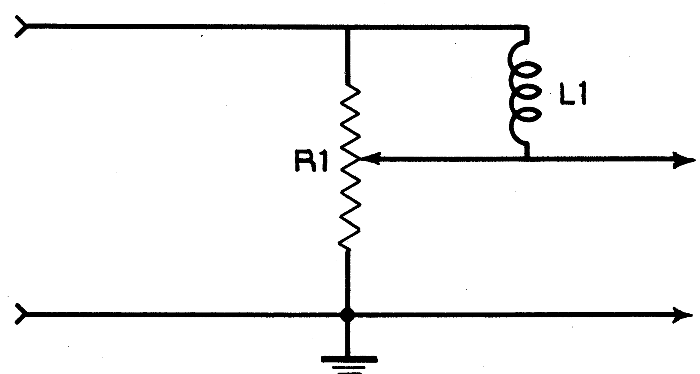
It has been indicated by Dr. Fletcher's work and borne out by tests with calibrated volume controls in observers' homes, that the average listening level in a quiet loca-



tion is approximately 70 d.b. above reference level or threshold of hearing. With this level as a basis and assuming that a desirable sound characteristic has been obtained at this level, it then becomes possible to plot a new series of curves from those shown in Figure #2 and which are shown in Figure #3. For convenience, the 70 d.b. level is shown as flat since we have already assumed that the subject level was satisfactory and thus any irregularities in the curve must be carried through to the other levels. From this family of curves, we can readily determine the amount of compensation necessary as the sound level is changed above and below the 70 d.b. average level. For instance, a decrease in average level of 10 d.b. at 1000 cycles must be compensated for by a decrease of only 5 d.b. in response at 100 cycles. Similarly, a decrease in average level of 20 d.b. at 1000 cycles must be compensated for by a decrease of only 10 d.b. at 100 cycles. The higher frequencies are not affected as much but it is desirable that the response be decreased only about 8 d.b. at 10,000 cycles for each 10 d.b. decrease in average level at 1000 cycles. Conversely, an increase of 10 d.b. in average level above the 70 d.b. level at 1000 cycles must be accompanied by an increase of only 6 d.b. in the 100 cycle level. An increase of only about 2 d.b. at 10,000 cycles should accompany this 10 d.b. increase in level. These curves may thus be used directly to determine the amount of compensation necessary at all levels above and below the average reference level of 70 d.b.

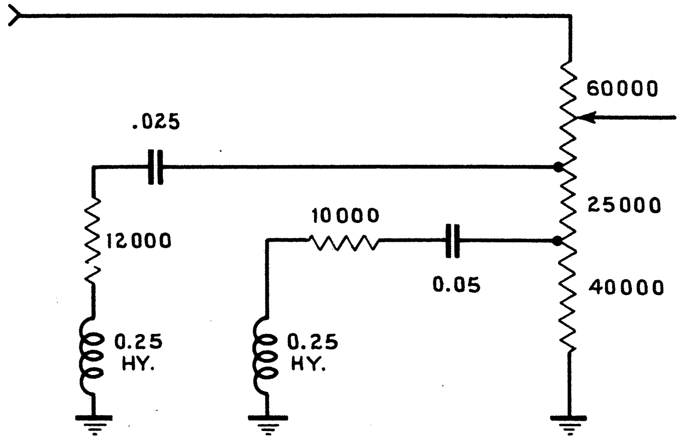
Having determined that some form of compensation is necessary, and with curves available showing the amount required for the different levels, we are faced with the requirement of finding ways and means of accomplishing the desired result.

One method which has been used in the past with good results on low impedance phonograph input systems is shown in Figure #4. In this, the volume control, which is of



SIMPLE COMPENSATION CIRCUIT
FIG. 4

the potentiometer type, has an inductor L_1 connected between the slider and the high end of the volume control. The size of the inductor will, of course, be determined by the resistance of the volume control with which it is used. In the particular case illustrated, the value of L_1 was approximately 30 millihenries and the volume control resistance was 60 ohms. With these values, the shunting effect of L_1 is almost negligible at 1000 cycles. At 100 cycles, and at mid point on the volume control, the shunting effect is approximately 2 to 1 or in other words, we have a rising characteristic 6 d.b. higher at 100



TYPICAL COMPENSATED VOLUME CONTROL
FIG. 5

cycles than at 1000 cycles at mid volume setting. This method of compensation is not adequate for better grade phonographs or radio receivers, but has been used with considerable success on lower priced instruments.

The next method, shown in Figure #5, has been found quite adequate for present day requirements. It will be noted in observing the diagram that there are two fixed taps on the resistance element in addition to the slider. These taps are used in conjunction with the proper shunting networks to give the proper amount of low and high frequency compensation for the different levels. The value of total resistance used is of necessity determined by the particular application. The positions of the taps are determined by the gain of the amplifier following the control and bear such a relation to the total resistance that normal listening level occurs at approximately the first tap down from maximum volume setting taking into account the compensation networks. The diagram indicates the values used for one particular compensated control which is being used in the instrument for demonstration. The shunt networks may have values changed as required to give the particular curve desired at normal level as well as the degree of compensation needed to fulfill the curves of Figure #3. A smaller capacitor will have the effect of increasing the low frequency response with reference to 1000 cycles whereas a larger capacitor will cause the opposite effect, other values remaining constant. The control and its attendant circuits are designed such that the change in level between taps is 20 d.b. When the slider has reached the second tap, no further compensation occurs. A further improvement can be shown by the addition of a third tap to the control, but this can be applied only to the higher priced instruments. The amount of high frequency compensation is determined by the size of the inductors used. Increasing the size of the inductor increases the high frequency response and conversely decreasing its size decreases the amount of tip up at the high frequency and as the volume is decreased.

A third type of tone compensation, which has been used in the past, is the automatic bass compensator. In this circuit, an amplifier system of the variable gain type is employed in parallel with the regular audio channel. By virtue of its variable gain characteristic and due to the fact that its response is limited to frequencies below 1000 cycles, it can be made to increase the overall low frequency response as the volume level is decreased. Due to the added cost and nicety of balance required for operation, this particular system has not had wide application in receivers enjoying quantity production. A similar arrangement has been applied to the radio receivers for restricting the amount of high frequency reproduction with a decrease in antenna signal voltage.

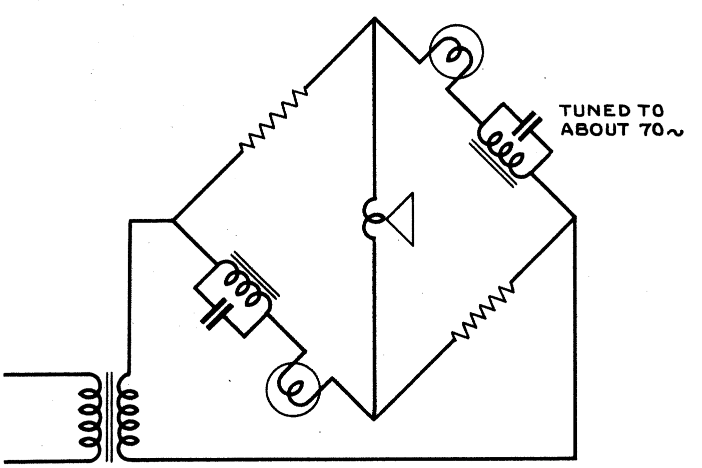
This covers briefly the subject of aural compensation and some possible methods of accomplishing the desired results. A short demonstration of this feature will follow as a means of showing aurally exactly what this means to you and me.

The next portion of this discussion will center around volume expansion as applied to both phonograph and radio reproduction. When volume expansion is mentioned to the average person, he immediately wants to know what advantage a set employing this system has over one which is not so equipped. To better understand the reasons for the existance of volume expansion, and to point out its advantages, let us consider changes in volume level of the orchestra, limitations in the reproduction of these changes as introduced by the phonograph or radio system and restoration of the original by means of an automatic volume expansion system.

By extensive tests, it has been well established that a large symphony orchestra has an available volume range of approximately 70 d.b. from a soft pianissimo to a heavy cresendo passage. If we were to try and cover this volume range on a phonograph record, we would immediately be faced with two definite limitations. The cresendo passage would, of necessity, require maximum movement of the cutter on the wax consistent with groove spacing and other mechanical limitations. With this as the top limit, the movement of the cutter on pianissimo passages would be so microscopic that when played back, the only apparent result would be surface noise or record scratch. Obviously, these passages must be brought above the surface noise level, in order to be heard and the cresendo passage must be limited as regard cutter travel, for the reasons outlined above. In doing this, compression of the 70 d.b. volume range occurs and we find a total volume range on latest records of approximately 50 d.b. When one considers that 20 d.b. difference represents a change in voltage or needle velocity of 10 to 1 and that 50 d.b. is 320:1 and 70 d.b. is 3200:1, I believe, it is readily apparent why the expression the director tried so hard to obtain during recording has been somewhat tempered by the time it is reproduced on an ordinary instrument. To illustrate by figures just what these limitations are mechanically on a phonograph record, let us consider that a recording is being made at about 100 grooves per inch. The maximum swing of the cutter point is thus limited to less than .010 inch if the grooves are not to touch each other. Assuming we could take this distance on loud passages, then the pianissimo passage, 70 d.b., lower than this would be approximately .000003 inch. This is much less than the microscopic structure of the record material itself and even an increase in the pianissimo passage of 20 d.b. still allows a swing of only .00003 inch. From these figures, I believe it is readily apparent that we are doing well to obtain the present volume range on the record alone.

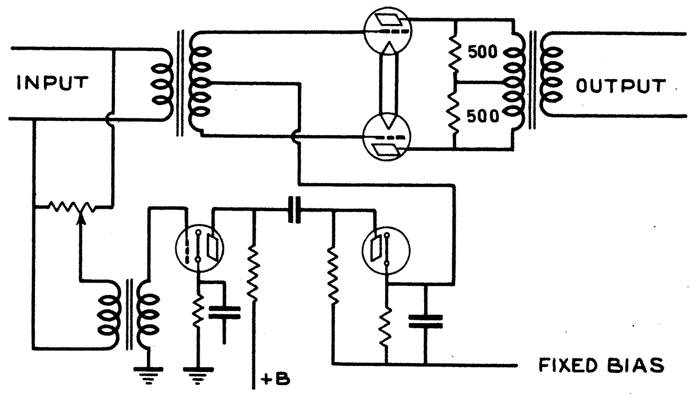
Limitations of a similar type are imposed by the broadcast station except for the fact that in this case line noises, hum and other extraneous disturbances require increasing the level on pianissimo passages and the danger of over modulation requires decreasing the level on cresendo passages. Present day high grade stations are able to cover a volume range of approximately 50 d.b. This is seldom used, however, since there is a definite desire on the part of the station owners and operators as well as program sponsors to obtain maximum listener coverage at all times by maintaining as high an average level of modulation as possible. Wide volume changes on present broadcast equipment would be satisfactory for people located close to the transmitter, but appreciation of it would be lacking by those located at some distance, particularly when the pianissimo passages were pushed aside by bursts of static.

The above discussion, I believe, indicates the desirability of obtaining some form of volume expansion as a means of restoring this loss of 20 d.b. in volume range particularly if full enjoyment of symphonic programs is to result. There are several ways of obtaining this result and the use of any one type is dependent upon the amount of expansion desired and cost permitted. With any of the systems which can be used at present, it is obvious that the result is a compromise since monitoring for all recording and broadcasting is accomplished manually and expansion in the phonograph or radio must take place automatically. Eventually, it may be possible to automatically compress the volume range during recording or broadcasting and so design the phonograph or receiver that its expansion curve is the counterpart of the compression curve. When this has been done, then the listener will be able to enjoy to a much greater extent, the various symphonic programs available. Until that time, however, we can obtain a great amount of pleasure even from the present volume expansion systems as applied to standard phonographs and broadcast receivers.



SYSTEM No. 1
BRIDGE TYPE EXPANDER CIRCUIT
FIG. 6

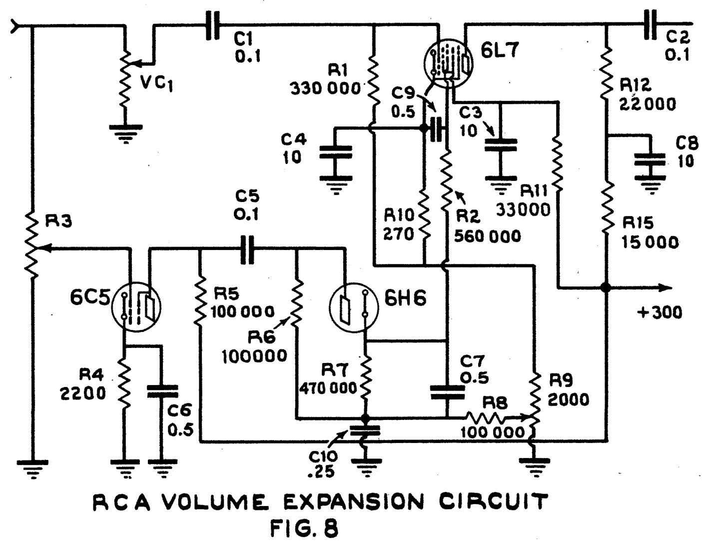
One relatively simple method, shown schematically in Figure #6, has been used the past year on a commercial broadcast receiver.[2] It employs low voltage electric lights of special design in a balanced bridge circuit across the output system. At low signal levels, the bridge is nearly in balance and very little signal gets through to the loudspeaker. As the audio signal increases, the resistance of the filaments in the lights changes and the bridge is thrown out of balance. This allows more than the direct increase in signal strength to be applied to the voice coil of the loudspeaker resulting in a degree of volume expansion dependent upon the strength of the applied signal. Earlier forms of this device imposed power output limitations upon the audio system but improvement in thermal resistance characteristics of the filaments used has resulted in a system capable of 15 d.b. expansion without serious distortion due to change in load impedance across the output tubes.



SYSTEM No. 2
PUSH PULL TRIODE, VARIABLE IMPEDANCE
EXPANSION CIRCUIT
FIG. 7

A second system, shown in Figure #7 consists of push-pull triodes having transformer coupling in both input and output circuits.[3] The plate circuit of the tubes, which normally calls for a load impedance of 20,000 ohms for proper matching, is shunted to approximately 1000 ohms by means of two 500 ohm resistors connected across the primaries of the coupling transformer. A variable bias system, dependent upon the strength of the incoming signal for its action, is connected to the common grid return of the triodes. In this type circuit, the transfer of energy between these tubes and the next audio stage is dependent upon the effective plate impedance of the tubes and the resistance of the load. Since the tube impedance is dependent upon the grid bias, there is a proportionally greater transfer of energy to the 500 ohm circuit at low grid bias than at high grid bias. In this way it is possible to use this circuit for volume expansion with very low distortion. Its main disadvantage, as far as commercial phonographs or broadcast receivers are concerned is its greater cost as compared with systems which do not require as many tubes and do not use transformer coupling.

The third system, shown schematically in Figure #8, has been used by RCA Victor for the past year for phonograph reproduction. Since this is the system to be demonstrated shortly, I believe it would be well to discuss the functions of the various parts. It will be noted that the incoming audio signal branches at VC_1. One branch



RCA VOLUME EXPANSION CIRCUIT
FIG. 8

goes to the #1 grid of the 6L7 variable gain amplifier tube and the other branch terminates in the degree of expansion control R_3. The 6L7 tube has previously been adjusted by means of variable resistor R_9 to operate on the proper portion of its characteristic for the amount of volume expansion and audio gain required. In the particular circuit, the tube gain is about 2.5 and the plate current is about 1 milliampere in the zero signal condition. As signal is applied and with R_3 set as maximum, amplification of the signal occurs in the 6C5 and rectification takes place in the 6H6. This rectified voltage, the value of which is determined by the strength of the incoming signal appears across R_7 and is thus impressed upon the #3 grid of the 6L7 through the time delay circuit composed of R_2 and C_9. The polarity of this voltage is opposite to that which is already present on the #3 grid and serves to reduce the effective voltage on this grid. This increases the voltage gain in the 6L7 and a variable gain amplifier results, the gain of which is entirely dependent upon the strength of the incoming signal. This system provides a relatively cheap volume expander capable of increasing the volume range 15-18 d.b. if so desired. The amount of expansion is easily varied by means of R_3 and in this way the desired result can be obtained on almost any type of phonograph or radio program.

A discussion of this subject must of necessity include some mention of the cabinet and its relation to the overall musical balance. The broadcast type cone loudspeaker radiates from both sides of the diaphragm. When this type speaker is mounted on a flat baffle and away from corners of the room or other cavities, very little low frequency resonance is present. When, however, the flat baffle is folded back to form a cabinet for the loudspeaker an immediate change takes place in the low frequency balance unless special precautions are taken. The cavity behind the loudspeaker cone serves directly in reinforcing the low frequency response and "boomy" reproduction results. To many people this type of reproduction is entirely pleasing and unless some "boom" is present they feel the set is not properly designed. To the music lover, however, this "boom" is highly objectionable since it is a type of musical balance, or unbalance, which never occurs in an orchestra.

There are many factors which govern the amount of cabinet resonance reproduced, among which the more important are: type of output system used; whether high or low impedance; frequency of resonance of cone suspension system; ruggedness and weight of wood used for the cabinet; depth of cabinet from speaker baffle to back opening and whether the back is open or closed. A brief discussion of each of these factors will enable us to understand more fully their direct effect upon reproduction.

If a high impedance output system is used, for instance one employing pentodes, changes in impedance of the plate load cause a proportional increase in voltage across the load due to the constant current characteristics of these tubes. A cone loudspeaker at its suspension resonance frequency presents a much higher impedance than at 400 cycles. For this reason it is desirable, if boominess is to be decreased, that the cone resonance be located below 70 cycles. Furthermore if a reduction in resonance voltage or output is desired at this frequency then a low impedance output system should be used. With either system the cone suspension resonance should never be located above 80 cycles in a console model since average cabinet resonance in this type cabinet occurs in the band between 100 and 150 cycles depending upon the cabinet depth.

Another factor directly connected with the amount of low frequency resonance effect is the weight of wood used for the cabinet. If thin woods are employed with very little bracing then at those frequencies where resonance occurs, or close to them, vibration of the cabinet sides results and undesirable responses occur. Heavy sides and bracing prevent this and as a result smoother reproduction of the low frequency portion of the music and voice range is obtained. If the depth of the cabinet is increased, cavity resonance occurs and even if the back is open there is an open organ pipe effect and undesirable responses result. For this reason it is highly desirable that the depth of the cabinet be restricted as much as possible consistent with good appearance. A back on the cabinet may or may not increase the resonance effect depending upon the cabinet design. In general the effect of adding a back will increase the undesired boominess unless special precautions are taken to acoustically ventilate the cavity.

There are many ways of overcoming this cavity effect to almost any desired extent, depending upon the additional cost of the apparatus. Some of these methods employ the back wave in the cabinet to advantage while others merely are concerned with getting rid of certain undesired effects of the back wave. In the absence of publications certain representative patents have been referred to where necessary for the technical material contained therein. One system for reducing cabinet resonance employs several speaker cones or other forms of diaphragms flexibly suspended in openings in the front of the cabinet. Early work on this arrangement was done by Mr. W. D. LaRue at the Victor Talking Machine Co. in Camden. Later developments have been made by Dr. H. F. Olson[4] at the RCA Mfg. Co.

Another system employs an acoustical labyrinth passage in the cabinet at the rear of the speaker for absorbing the back wave without undesired reaction upon the low frequency response of the speaker. In one form of apparatus the exit of the labyrinth has been employed to reenforce the low frequency waves although, in such a case, best results have been obtained by making the labyrinth expand exponentially, thereby constituting a folded horn loading the rear of the diaphragm. Early work was done on the labyrinth acoustic baffle by Mr. Julian High[5] at Westinghouse Mfg. Co. Later work with a labyrinth baffle of the horn type loading the rear of the diaphragm has been done by Dr. H. F. Olson[6] in connection with high fidelity theatre installations and broadcasting monitoring speakers.

Still another system for overcoming cabinet resonance has employed one or more absorption chambers or wave traps tuned to the frequencies of troublesome resonant peaks. Early work on this arrangement was done by Carlisle of Westinghouse and later developments have been made by Dr. Irving Wolff[7] at the RCA Mfg. Co.

Another system being used this season employs a solid back on the cabinet and a very solid type of cabinet construction. Acoustic ventilation and re-enforcement of the low frequency end of the music and voice range is obtained by a series of pipes located in openings in the bottom of the cabinet. By determining the size and number of these pipes for a given cabinet, it is possible to extend the low frequency response of the over-all sound output one-half to three-fourths of an octave and at the same time to reduce the response from six to nine d.b. at the low frequencies of the voice range, around 100 to 120 cycles. Reference is made to developments by Thuras[8] at the Bell Telephone Laboratories, and to more recent work by C. O. Caulton of the RCA Mfg. Co.

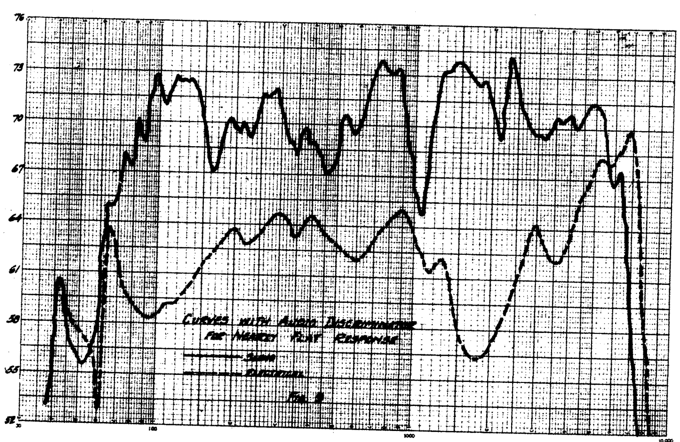
Further refinements for improving acoustic reproduction consists in an inclined speaker baffle in a cabinet. C. R. Garrett[9] and I did some early work on this for the purpose of reducing cabinet resonance. Another refinement consists in a high frequency beam spreader in front of the speaker cone, developed by Dr. Irving Wolff. For high fidelity work the double voice coil speaker, developed by Ringel and Olson, has been used for extending the high end of the range to 8,000 and 10,000 cycles.

In the absence of technical publications on the above material, reference has been made to patents for convenient reference by those interested in obtaining further details.

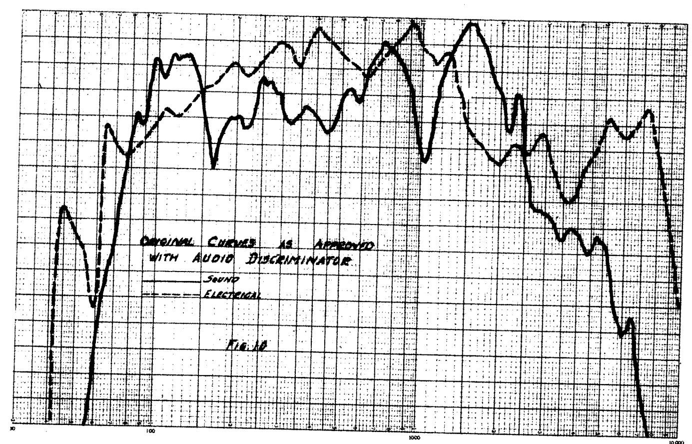
The next portion of the paper will deal with a piece of laboratory equipment which has been very helpful in determining the desired audio frequency characteristics for a given amplifier or input system to obtain pleasing sound output. It is called the audio frequency discriminator. Fundamentally, this device is a compound filter and amplifier provided with a system of controls which permit its frequency characteristics to be altered to almost any desired extent. This control of the frequency characteristic is effected by a division of the audio range of the amplifier, namely 20-10,000 cycles, into eleven filter bands the gain in each band being individually under control. The bands overlap at the sides and are so phased at these points that the combined overall response may be made substantially flat if so desired. The individual bands are slightly less than one octave in width at the overlap point and have a range of amplitude control averaging 12 d.b. up and down from the flat characteristic. A switch is provided which permits the operator to quickly transfer from the normal audio system to that which incorporates the discriminator. In this way it is very easy to compare an audio system which is being worked on with one having the desired characteristic and in this way determine the changes that are necessary to correct the former.

In addition to the eleven filter stages, the discriminator is provided with a continously variable high frequency cutoff filter. This filter is in no way connected with the band filters and allows a much finer control of the upper limit of the tonal range than does the band filter. It has essentially a vertical cutoff over its entire range from 3500 to 10,000 cycles. This cutoff filter may be switched in and out of the circuit as desired.

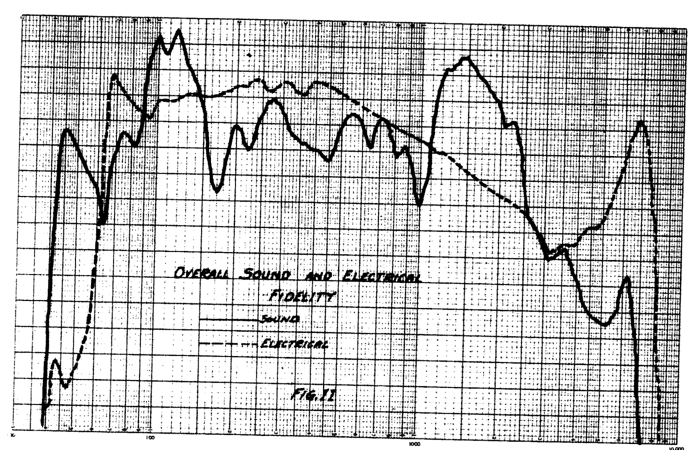
Figure #9 shows the overall electrical and sound curves of the phonograph being demonstrated with the discriminator adjusted for essentially flat response from 60 to



7,000 cycles. Listening tests on this instrument with this setting indicated that surface noise from a commercial standpoint was highly objectionable. As a result the discriminator controls were readjusted to give the curves shown in Figure #10. It will be noted that while



the range remains essentially the same there is a definite tendency toward a trailing off characteristic above 4,000 cycles. The present recording system employed in Victor records has a slightly rising characteristic in this range and the overall result is one which is very pleasing from a musical standpoint; yet the surface noise is not objectionable. Having determined the desired characteristics, it then was necessary to provide the proper equalizing network to obtain similar performance without the



discriminator. Figure #11 shows how closely it was possible to duplicate results. A brief demonstration of the use of the discriminator will be given at this point.

B I B L I O G R A P H Y

(See next page)

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HIGH FIDELITY RADIO RECEPTION

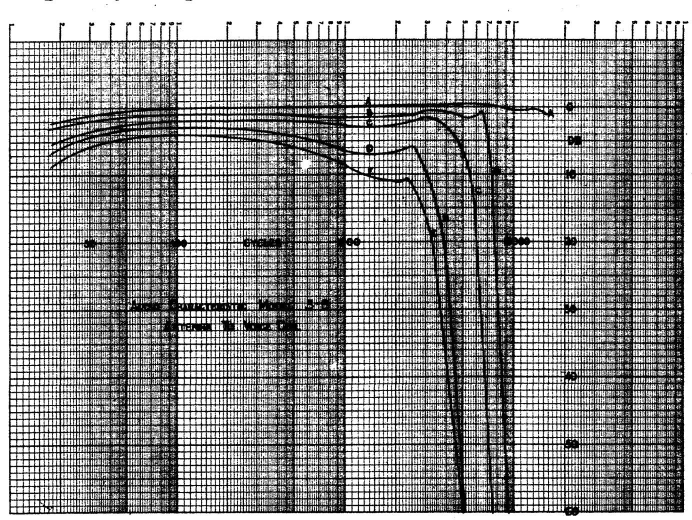
BY

LINCOLN WALSH*

Delivered before the Radio Club of America
December 10, 1936

A high fidelity system might be defined as a system of picking up sounds, transmitting them, and reproducing them so that they sound to the ear precisely like the original sounds. It can also be defined as a system which picks up the sounds, transmits and reproduces them, with all the original sinusoidal components present in their original proportion and phase relation but with the introduction of no new components. If the system does this, the reproduced sound duplicates the original and the system is an ideal high fidelity system. There remains, then the question of how closely a practical system must approach this ideal in order to be satisfactory and this can be answered only by the ear.

Perhaps more than any other sense, the sense of hearing is influenced by suggestion. If the listener has in his mind the conviction that his radio gives him an exact reproduction of the sound in the studio, his ear tells him that there is nothing wrong with the fidelity of his radio. But while the ear of such a listener tells him that all is well with the sound to which he is listening, even when he listens critically to judge the quality of reproduction, an inquiring observer would notice that the extent to which he listens casually to his radio bears a direct relation to its quality of reproduction. Actually, a person will tire quickly of listening to a receiver of poor tone quality, and yet not be conscious of its poor quality. Conversely the same person listening to a receiver of good quality may not be conscious that the quality is better, yet he will listen to it for much longer periods without tiring. It is a matter of common observation that there are many homes having midget receivers of obviously poor quality, whose owners are very proud of the quality and performance of their receivers, and yet those receivers are turned on only for special programs while in homes having receivers of better quality they operate almost continuously. Thus, while the ear is a highly uncritical instrument, it quickly tires of listening to voice and music which is distorted or lacking in important frequency ranges.



The musical instruments of today have evolved thru centuries of listening, and the test which determined their survival was whether or not the tone was suitable as judged by the ear. These centuries of experience have shown that sounds of all frequencies thruout the wide frequency band of audibility are requisite for musical expression and so, when we as radio engineers design radio broadcasting receivers so that they cut off or seriously attenuate frequencies below about 100 cycles and frequencies above 4500 cycles, we are not only undertaking to give a new interpretation to music but we fly in the very face of man's centuries of musical experience.

The human voice is one sound to which we are always listening, and this experience therefore provides an excellent basis for the ready evaluation of the fidelity of reproduction. Even a receiver of very poor fidelity gives a high degree of understandability, because, as is well known, a range of 300 to 3000 cycles is all that is necessary to give understandability. But to give the naturalness that is necessary if the reproduction is to be untiring to the listener, it is necessary to reproduce the entire audible range.

A particular instance of the importance of this is the reproduction of soprano voice. We all know how often the re-

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production of the voice of a soprano as heard over the radio is lacking in the qualities that make it a pleasure to listen to the singer in person. This results from the fact that the fundamental frequency is usually far higher than that of average speech, and the harmonics which give the voice its "color" are not reproduced by the receiver. Yet on a high fidelity system all the overtones are present and the reproduction is natural and pleasant.

It is a fact, however, that sometimes a program sounds better if the highs are reduced by lowering the cutoff at the high end. This is invariably due to the presence of distortion or to a high background noise level. In the absence of distortion and noise, any normal ear will choose the highest available cutoff.

The ear like most human senses is subject to habit, and if a person is accustomed by habit to listening to a receiver with a low cutoff, he may not immediately react favorably to a high fidelity system. A receiver having a medium cutoff and a peak near that cutoff, may at first sound to such a person as if it has more highs than a truly high fidelity system, but, again, he will be found to tire quickly of listening to the receiver with a peak. But he will listen indefinitely to the high fidelity system without fatigue.

A listener in a comparison test between a high fidelity system, and a system having a lower cutoff, will very often choose the lower cutoff at first but if he takes time, sometimes as much as an hour or two of listening, he will inevitably choose the high fidelity system. Some listeners when first hearing a high fidelity system by itself think it tinny, some think it very bass, some think it has too much bass and too much treble, and lacking in middle register, because they are hearing tones they are not accustomed to hear in radio reproduction. But after listening for an extended period to good program material they like it. And that is the final, and the only reliable test.

These observations are the result of a systematic study of listener reaction started by the writer some nine years ago. They report the conclusions arrived at after a study of the reactions of something over one hundred listeners only a relatively few of which were radio engineers or serious students of music. On the basis of these observations the writer long ago undertook to develop receiver design details to supply the latent and all too little recognized desire for a much closer approach to complete fidelity in radio reproduction. The results of that work have been incorporated in a receiver typical of the especial arrangements which have been found necessary to meet this need and the performance of that receiver will be demonstrated. Before proceeding to the demonstration, however, it will doubtless be of interest to describe something of the design details and their specific purposes and functions.

THE HIGH FIDELITY RADIO RECEIVER

From the standpoint of present broadcast receiver design practise, high fidelity means extending the audio frequency range at both its ends. It has long been known that to have the tone balanced and most pleasant, the audio range of a system must be centered somewhere between 400 and 1000 cycles. If we extend one end of the range, the other must also be extended for best effect.

THE LOUD SPEAKER

At the low end, it has been found desirable to extend the range of the amplifier to below 30 cycles, not withstanding the limited effectiveness of commonly available speakers in that range. To assist the speaker in this range the largest possible baffle must be used. Olney's work on accoustic labyrinths has pointed the way to improve low frequency speaker response where only limited baffle area is available. An electrical resonance in the circuits, or a mechanical resonance in the speaker - more commonly the latter - is sometimes used to provide a peak at about 100 to 130 cycles, which results in the "boom" of false bass response. In any program, there is always enough energy in the region of this peak to give the low pitched background which many consider to show good bass response. But this causes listener fatigue to develop very quickly and it is thus extremely undesirable to improve bass response by means of any such resonance, either electrical or mechanical. If speaker resonance must be countenanced it should occur at the lowest possible frequency, certainly below 50 cycles; the amplifier must be flat within 2 db down to 30 cycles; the baffle must be of corresponding size, or a suitable labyrinth must be employed; and any bass compensation that is employed must be entirely of resistance-capacity type, to avoid any bass resonance.

THE 'TWEETER'

For the high frequency portion of the audio range an especially built "tweeter", which is quite similar to a standard 6 inch cone speaker has been found to supply the best practical solution to the problem presented by the need for efficient translation into acoustic energy of the high audio frequencies. Such a "tweeter" has been included in the demonstration receiver. It has a good sound pressure response curve up to 14,000 cycles, and it is not seriously down at 16,000 cycles. This is secured thru the use of an extremely light voice coil, a short tube connecting the voice coil to the cone, and a light paper cone, of rather low damping.

The "tweeter" is connected thru a small condenser directly across the plates of the push-pull output tubes, so that it cuts in gradually above 2000 cycles. Experience indicates that it is better to have a gradual transition from the low frequency speaker to the tweeter, than any abrupt transition as results from the use of sharp cutoff filters.

While speakers can be built that respond well up to 9000 cycles, the combination of a low frequency speaker and a "tweeter" shows itself to be far superior to any single speaker. One reason for this doubtless resides in the fact that the motion of a large cone diaphragm at high frequencies is made up of two sets of waves radiating out from the voice coil. One wave is longitudinal with respect to the paper of the cone while the other is a lateral wave in the paper, the former travelling at considerably higher velocity that the latter. Their propagation in the paper and their reflection at the edge of the cone determines the high frequency response, and the control of all of these factors is a far more complex and difficult problem than making two speakers of distinctly different proportions each with definite and supplementary characteristics. Additionally where efficient response up to 16,000 cycles is desired no single practical speaker has been found to serve at all satisfactorily.

ELIMINATION OF DISTORTION

Distortion is a very important factor about which volumes could be written. The problem starts at the RF amplifier. The signal voltages at the grid of this tube must be held low to prevent overload and harmonics that will modulate the carrier.

The converter in the demonstration receiver is a special circuit which has a very low noise level. The detector delivers an audio voltage only, and has no relation to the AVC system which is separate. The automatic volume control system holds the detector input voltage at 10 volts for all normal signal inputs, in order to avoid the distortion which occurs in a diode when operated at low voltage levels - within the "parabolic" range - and to avoid distortion in the last IF tube due to overload, which would occur if the diode had to be driven to high voltages.

Additionally, of course, the AVC system holds signal voltages thruout the receiver at such values that will avoid overloading any of the tubes such as the converter and the IF tubes.

It is not, perhaps, commonly appreciated to what degree the use of silicon steel in the interstage audio transformers introduces harmonics, particularly at low levels. But because of this fact the audio signal is carried thru resistance capacity coupled circuits, having no iron core devices anywhere, up to the input of the push-pull output tubes. At this point a push-pull transformer of special design, including a core of high-permeability alloy, which does not generate harmonics is employed, and which contributes greatly to the clean tone quality of the receiver.

All the audio amplification is provided by the use of low-mu triodes, which are the only amplifiers sufficiently free of distortion for a high fidelity system except, perhaps, as the newly developed degenerative circuit arrangements may make the multi element tube less unsuited to this field.

THE I. F. AMPLIFIER

The problems presented by the need for so designing the selective high frequency amplifier stages as to provide for high fidelity reception are basically impossible of solution since, under the American scheme of broadcast frequency allocation and assignment in which adjacent assignments differ by only ten K.C. and practically all frequencies so assigned are in simultaneous use, the requirement that interference-free reception be possible on any assigned frequency at any time and place, unavoidably limits the audio band width of reception to something less than five thousand cycles. Under these limitations there is little that the radio designer can do other than to provide a relatively narrow band width and pray that it will be found not too unacceptable. And, indeed this is precisely the direction in which the receiver designs of recent years have gone.

It is patently absurd, however, to so limit the fidelity of receivers so that only such painfully low fidelity is available to the listener who is located relatively close to his local transmitter or who may be in the high field strength area of a high powered transmitter and thus largely free of adjacent channel interference and interference from noise sources. And since the system under which our radio receivers are distributed to the purchasing public requires so complete a universality of usefulness there is obviously no solution but to give the receiver such variable selectivity as to provide, on the one hand, so narrow a band width as will allow of distance reception in areas of noise and interference and, on the other hand, to provide so great a band width as will allow of the reproduction of the entire audio range being broadcast by the best transmitting system.

In the demonstration receiver this is accomplished in the intermediate frequency amplifier since, as is usual in the superheterodyne type of receiver here largely resides the selectivity of the system. It has been found best to use two IF stages, including three double tuned IF transformers, the coupling between the primary and secondary circuits being varied by moving the secondary coil relative to the primary. At the position of minimum coupling, the coupling is considerably below critical, and the selectivity is at its highest. At the position of maximum coupling, position A in the figure, the resonance curve of the first two transformers becomes double peaked, with the peaks separated by about 35 kilocycles. The third transformer which feeds the detector is broadened, but not enough to show double peaks. The single peaked resonance curve of this latter transformer fills in the valley of the combined curves of the other transformers so that the resultant overall curve of the IF system is flat over a band of about 32 KC, thus permitting the unattenuated passage of sidebands corresponding to all audio frequencies up to 16,000 cycles. A second step of coupling, position B, is provided in the receiver in which similar conditions exist, with, however the band width reduced to about 16 KC corresponding to an audio band of 8,000 cycles. Positions C and D have band widths of 12 and 8 KC respectively, and the fifth position, E, is the position of minimum coupling, and passes about 5 KC. The audio bands corresponding to these are, respectively, 6,000, 4,000, and 2,500 cycles.

It might be well to point out in passing that the use of variable inductive coupling as here employed has certain advantages over other possible types of coupling that might be employed, wholly aside from the obvious advantages of economy of production, ease of production adjustment etc. It will, of course, be remembered that, in general, the peaks in the transmission characteristic of a pair of tuned and over-coupled circuits can be equal only when there is no loss in the coupling element and since mutual inductance is the one coupling element that is loss free, it is thus especially suited to this purpose. Another factor of interest here is that by varying only the mutual by the motion of one of the coils, the band width is varied while maintaining the midfrequency fixed.

R. F. AMPLIFIER

The radio frequency system has as its primary function the elimination of the image frequency, which in the demonstration receiver is 940 KC higher than the signal frequency. It serves also to eliminate such other signal frequencies, as might be brought in thru beating with harmonics of the oscillator. The RF system may, therefore be made broad enough to suit the widest band passed by the I. F. amplifier and need not be of variable band width. The antenna circuit of the demonstration receiver is double tuned, and double peaked, with peaks separated about 35 KC. The RF amplifier is single tuned and is broadened by the use of very fine wire in the winding, so that it just about fills in the valley of the double peaked antenna system. At the higher end of the broadcast band, the valley of the antenna circuit is less deep, and the RF stage is less sharp, so that the RF system as a whole is flat within 1 db over a band of about 35 KC.

This gives a system which will pass all frequencies up to 16,000 without any attenuation. When this system is tried on the air, the tone quality is all that might be expected. But after sunset, when the distant stations on adjacent channels begin to come in, 10 KC whistles are heard. If a filter is put in circuit to cut out 10 KC, the whistle disappears, but then the so called "monkey chatter" is heard. This chatter is the high pitched unintelligible sound which is due to the side bands of the adjacent channel beating with the carrier of the desired signal, in contradistinction to the more commonly experienced interference, known as "cross talk" which is the result of the adjacent channel side bands beating with their own carrier. It is of interest to note that in the receivers which have been built on the basis here discussed, crosstalk from the adjacent channel has been far less serious in creating interference than the adjacent channel chatter. As a matter of experience, it has been found that chatter is the ultimate limitation on fidelity.

It has been found however that, when receiving moderately strong signals from local stations, practically all the chatter can be eliminated by the use of an extremely sharp low pass filter, having its cutoff at about 7500 cycles.

The reason for this is not hard to find if it is remembered that workers in the acoustics and telephone fields have shown that most of the energy in speech and music lies in the frequency range below 2000 cycles, and that the energy drops progressively as the frequency under inspection increases. Thus it is found that there is very little energy in the region of 8000 cycles and higher although that little energy is of great importance in giving naturalness to the sound of which it is a part. Similarly, then most of the energy in the sidebands of a signal is in those frequencies differing by less than two thousand cycles from their carrier and hence by more than 8000 cycles from the carrier of the adjacent channel transmission. Thus, the resulting "monkey chatter" carries most of its energy in the frequencies above 8000 cycles and is thus subject to effective elimination thru the suppression of all frequencies of that order or higher.

THE AUDIO FILTER

To eliminate both the chatter and the whistle in the demonstration receiver, there is used a 4 section Campbell type filter, having 11 elements; 4 series inductances, 5 shunt capacities, and 2 mutual inductances. This filter is flat up to 7200 cycles, and is down approximately 30 db at 8000, 50 db at 9000, and 70 db at 10,000 cycles. These values of attenuation have been found necessary to receive local signals without chatter in the 7500 cycle setting of the fidelity control.

By means of the fidelity control in the demonstration receiver, the band width of the I.F. is varied and thru a simple mechanical linkage with the audio filter, the cutoff of the audio system is simultaneously changed to provide a cut off just below the cutoff of the IF system. This gives the highest possible audio band width for a given selectivity.

In practical use it has been found possible to receive signals as low as 10 millivolts without chatter on the 7500 cycle setting, but with all the superior fidelity implied by that high cutoff as compared with conventional receivers. Weaker signals may require that the cutoff be set at 5500 cycles, which provides better than usual fidelity. When very high selectivity is desired as in cases of high noise levels and in the reception of distant stations, the band is narrowed to 4000 or even to 2500 cycles. This latter band width appears pointlessly narrow but it must be remembered that many listeners want high selectivity and the ability to get distant signals for a short period after they buy their receiver and these circuit arrangements make that possible. Happily, however, after they experience the pleasure of good programs they want little more than the nearby stations, and these they may receive on the high fidelity ranges free of chatter or noise and with highest fidelity consistent with the conditions of transmission and reception.

Experience indicates that in daylight the local stations can usually be heard without whistles, chatter, or any noise on the 16,000 cycle setting, and indeed many stations are transmitting programs of a quality which shows a definite improvement when the cutoff is raised from the 7500 to the 16,000 cycle position. Especially well does this type of receiver operate when receiving the high fidelity stations which have 20 KC channel separation, and good lines and amplifiers, and which are therefore best listened to on the widest band.

FIDELITY OF BROADCAST TRANSMITTERS

It is often argued that the broadcasting stations themselves do not supply programs with such frequency band widths as to justify the use of high fidelity receivers. With this in mind the writer sometime ago made casual investigation as to the upper cut-off of the higher powered transmitters in the New York area and he was happily surprised to note the care which has been taken in the design of much of the broadcasting equipment to maintain the upper frequency limit of the equipment at such a high value as to provide for truly high fidelity transmission. It was found on inquiry, for example, that the studio and transmitter equipment of WJZ and WEAF is good up to 16,000 cycles per second as is that of WABC and WOR. The telephone lines connecting the studios and the transmitters of these several stations are reportedly not as good as the equipment. Thus, the line to the WEAF transmitter is reported to cut off at 7000, the WABC line, at 8500, the WJZ line at 10,000 and the WOR line at 11,000. These latter data on the cutoff frequency of the telephone lines are not to be viewed as inherently limiting the broadcasting system since, they doubtless result largely from economic and not technical factors, and can and doubtless will be raised as the demand for such improvement developes. It must, however, be admitted that there is little incentive for any broadcaster to assume the added burdens of cost and maintainance required by any further expansion of the band width of his transmission until and unless the receivers in the audience to his transmissions are capable of taking full advantage of that transmission.

Thus, once again we have arrived at the stage in the development of broadcasting where the transmitter leads the receiver and now awaits being overtaken. Thruout the history of broadcasting, first the receiver and then the transmitter has been the more thoroughly developed element of the system and in each step in the sixteen years of progress in radio broadcasting each improvement in the lagging element has placed it so markedly in advance of the other as to provide the incentive for major improvement in the other which in turn prompted further improvement in the first, and so on and so on around the widening spiral of progress.

It seems quite reasonable, therefore, to believe that not only will the next move in this continued progress be made by the receiver designer-in the direction here described in detail - and with the agreement of his commercially minded associates, of course - but that that next step will leave the broadcaster lagging once again only, however, to have him soon moving again in the direction of the ultimate perfection of broadcasting.

EDITOR'S NOTE

In the course of the delivery of Mr. Walsh's paper demonstrations were made in connection with a special program for the Club from radio station WQXR under the direction of Mr. John V. L. Hogan. This program included a variety of broadcasting material designed to show the superior effectiveness of the high fidelity reception through the transmission of a wide range of tones and special musical transmissions both with and without the use of cut off filters at the transmitting station. It can be reported that any cut off at the transmitter that tended to reduce its transmission band width below the maximum possible was quite easily evident in the reception as heard by those in attendance and it was of especial interest for most of the members of the club present to make comparisons between the reproduction of Mr. Hogan's voice and their memory of it as it so often when Mr. Hogan attends in person. The demonstration left no doubt as to the need for the transmission of the entire band width of the receiver and the transmitter where completely satisfactory fidelity is the aim. Additionally it should be pointed out that while this demonstration was made at Havemeter Hall, Columbia University at which previous experience has shown that the noise level is always objectionably high, the installation of a noise suppression antenna system, through the kindness of Messrs Amy and Aceves and King, left little noise interference to detract from the demonstration or from the enjoyment of the special musical program that was transmitted.

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