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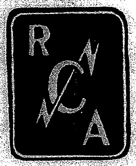

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DESIGN AND APPLICATION OF ADJUSTABLE TONE COMPENSATING CIRCUITS FOR THE IMPROVEMENT OF AUDIO AMPLIFIERS†

By JULIUS G. ACEVES*

THERE are a number of problems or questions which will arise after pondering for a while upon the intricacies of the processes for both recording and reproducing sound waves.

In the first place, the sound waves with which we deal are very complex in shape; very seldom we find pure sine waves in music and never in speech. This being so, how can we understand the operations of the various apparatuses and their results in view of the fact that in our theories we assume sine waves? A French scientist, while investigating heat conductivity, in 1822 in his "*Theorie de la Chaleur*," gave to posterity the key to the solution; J. B. J. Fourier developed a theorem that applied to problems of the nature that we are discussing, and which will furnish the basis of the analysis. This theorem proves that any recurrent curve can be expressed by an equation containing a sum of sines and cosines and is well known to radio engineers. In ordinary language it means that any periodic curve can be expressed by a sum of sines of terms involving a fundamental frequency and its multiples. Consequently, any periodic wave can be studied by analyzing the effects of the individual components. There are quite numerous cases, however, where there is no repeated cycle. Such cases may be analyzed by decomposing the curve into an infinite number of frequencies infinitely close together like a continuous spectrum, as may be shown by Fourier's Integral. In either case, however, so long as in the final sound wave, the coefficients affecting each of the component frequencies are kept identical or at least in the same ratio with respect to each other, as in the original sound wave that was recorded, there will be absolute fidelity even if, during the process, these coefficients lose their relative proportionality, and this seems inevitable. Therefore, there is an absolute necessity of compensating for the inequalities in the way that the coefficients (or maximum amplitudes) of the various frequency terms in the expression of the complex wave will suffer.

Let us state again: in order that a wave which is decomposed into its simple harmonic components will keep its shape, it is necessary and sufficient that each of the terms of the series (or integral) has the same or proportional coefficients that are in the original, and that the phase relationship be preserved between all the components. This is exemplified in the case of submarine telegraphy, where it is essential that the original wave shape of the sending signal shall be reproduced at the distant end in spite of all the distortion of magnitude and phase that takes place in a trans-oceanic cable. This distortion has to be compensated *both* in magnitude and phase before the received signal may begin to look like the original sufficiently well so that it may be interpreted.

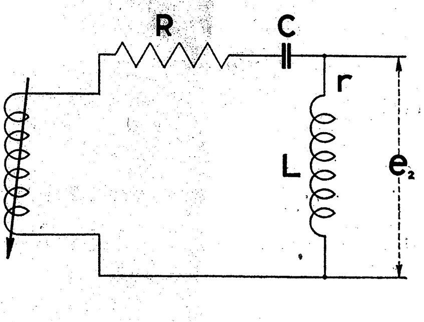

FIG. 1

Schematic circuit of one stage of amplification.

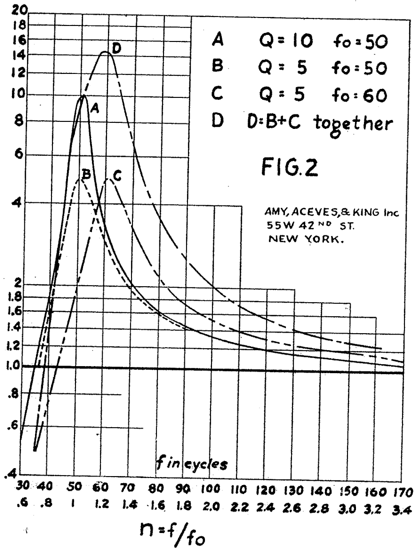

Fig. 2
Values of gain for various frequencies.

In the case of sound, it is very fortunate that the phase distortion does not affect the ear except under very rare conditions. Hence, for a faithful reproduction of recorded sound, we must have a constant relationship between the coefficients of all the frequencies at the reproduced air wave, as in the original. But right here is a difficulty: If we take for example just two terms of a complex recorded wave of the form:

$$Y = A\sin(\omega t + \Phi) + B\sin(N\omega t + \Phi_n)$$

where Y represents the lateral displacement of the needle in the groove, the voltage induced in the pickup will be represented by the derivative of this equation since $E = A\dfrac{d\Phi}{dt}$ where Φ is the flux interlinking the coil of the pickup. We shall find that the derivative of the term $B\sin(N\omega t + \Phi)$ has a coefficient N times as large as that of the derivative of the fundamental, or $A\sin(\omega t + \Phi)$. If we examine some of the photographs of recordings shown by Professor Powers[1] it will be seen that this is likely to be so in practice, since for about the same intensity of reproduction of a bass note, the indentations in the groove of the record are much larger than for a high note, and yet they sound about equally loud. Then again the question arises—what are the characteristics of the loudspeaker, assuming that it receives an

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† *Delivered before the Club at Atlantic City on June 3, 1930.*

* *Chief Engineer, Amy, Aceves & King, Inc.*

[1] *Proceedings, Radio Club of America, Vol. 7, No. 4.*

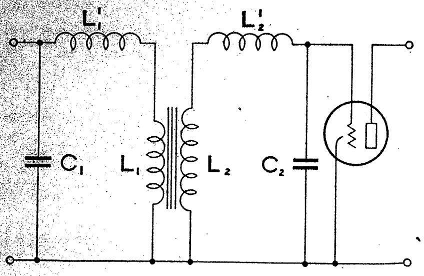


FIG 3

Equivalent circuit.

undistorted amplified wave from the pickup across its terminals?

It is very difficult to offer a mathematical solution to these problems a priori, without the aid of experiments, because there are many characteristics which are too complex to express by simple mathematical expressions, such as the relation between lateral motion of the cutting tool that made the record, and the applied e.m.f. to the terminals of its driving coil at various frequencies; the impedance of the loudspeakers vary not only with the frequency but with the acoustical conditions of the surrounding medium; the free vibrations in the mechanical and electrical circuits which take place besides the impressed or forced vibrations at certain frequencies, and other innumerable non-linear links in the whole chain, the theoretical performance of which is only an approximation with plenty of limitations.

One consideration alone "saves the day" in so far as the theoretical treatment of the problem is concerned, and that is, that the derivatives of a sine are always sinusoidal and so long as we can decompose complex waves by means of Fourier's theorem into sine components, and that the phase relationship between them does not matter for the reproduction of sound as it affects the ear, all we have to do is to modify the coefficients of the various "partials" in such manner that the last wave in the chain of transformations, which is the air pressure variations with time resulting from the motion of the loudspeaker diaphragm, shall have components with coefficients nearly proportional to those in the original air pressure wave.

It is fortunate that the amplifiers used in the recording and reproducing operations are susceptible of affecting the coefficients of the various frequency terms in the whole series of integrals. By the introduction of compensating circuits, it is possible to reinforce certain frequencies and subdue some other ones at will with a minimum of free vibrations introduced thereby.

Although, from a theoretical standpoint, there is no limitation to the way in which the coefficients of the various terms may be modified, practical considerations will narrow these limits considerably as we shall see in our present discussion.

In order to see more definitely the scope of our problem, let us glance over, rapidly, the progress of sound recording and reproducing in the last few years.

Since the invention of the phonograph until the time when the vacuum tube was developed more as an amplifying device than a radio instrument, the acoustic limitations of recorded music were so great that it is a wonder that people could listen to such musical monstrosities as we had in those days. In the matter of vocal music and speech, conditions were not half so bad as in the case of instrumental music because the portion of the scale that was recorded and reproduced fell almost within the range of the human voice. Tones below middle A (435 Int. Pitch) were heard almost only in their partials. Above A^2 (1340 c.p.s.) there was very little; consequently the articulation of such consonants as "S", "F", and "V" was very poor, and recorded violin tunes near that pitch could not be distinguished from a flute playing the same notes.

With vacuum tube amplifiers and broadcast microphones, it was possible to obtain records with a wider range in pitch which, when mechanically reproduced through diaphragms in the end of long exponential horns, sounded much better than the former records. Here we find fundamental tones of 200 c.p.s. and notes as high as F (2760 Int. Pitch). The absence of fundamental tones in the 50-100 and 100-200 cycle octaves deprived the music of any majesty of effect and the depth of sonority that are the glory of the organ and the orchestra. From then on, these low tones have been gradually introduced in records, and the last six months have seen a tremendous improvement in the quality of music from commercial phonograph discs, not only in classical music but also in dance music.

Many tests have been conducted to determine the effects of the suppression of given bands of frequencies from the musical scale while carefully listening to a variety of renditions. The Bell System Technical Journal may be consulted for information on this subject, and Dr. Fletcher has done a great deal of work not only in the effects of frequency suppression in music but also in speech and intelligibility of articulation on one side, as well as in naturalness on the other.

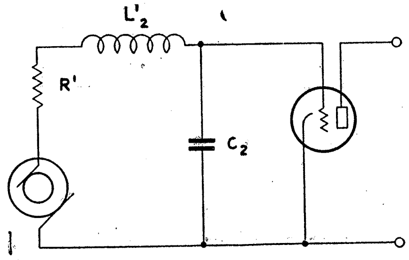


FIG. 4

Variation of circuit shown in Fig. 3.

For the purpose of obtaining a mental picture of the two characteristics of pure tone, namely—pitch and intensity, we have arranged a short demonstration. First, we shall get acquainted with the sensations of tone produced by pure simple harmonic tones as given by the loudspeaker actuated from an oscillator. Here we must take into account the fact that even if we had a wave in which the harmonics were absent, at least to 1/1000 part of the fundamental, the loudspeaker would introduce harmonics due to the non-linear relations between propelling forces and corresponding displacements, although for the purposes of the demonstration they may be considered negligible.

Having refreshed our recollections of pitch, we will now see what are our impressions of intensity. The unit of sound intensity which is commonly employed in engineering work is the decibel. As applied to an electrical circuit, it is defined as ten times the common logarithm of the ratio between two energies; as a rule between energy entering a network and energy coming out at the end. If the logarithm is positive, the output is greater and the circuit is said to have a "gain" of so many decibels. It implies the presence of an amplifier within the network.

As in all quantities involving a ratio, the logarithm of the ratio is the difference between the logarithms of the numerator and denominator. Consequently, when we say that there is a gain of 8 decibels, we simply mean that the energy "level" is higher by an amount of 8 decibels above a level which we take as reference. This is the same as to say that the output energy is about 6¼ times as great as the input, whatever the absolute value of this input may be. In practice, the softest audible sound is taken as "zero" level and the loudest sound, which is so loud that it actually causes pain, is about the sensation of one hundred decibels above the zero level. As the sensation of loudness appears to be a logarithmic function, a gain of 10 decibels, from zero to ten, is felt about as considerable as a change from 40 to 50. Hence, we are going to choose an arbitrary level and see the effect of increasing the tone energy by plus or minus ten or more decibels. Here it should be noted that the energy levels from the two thresholds; that of inaudibility, and that of pain caused by excessive loudness, are closer together at the extreme ranges of the audible spectrum, and for this reason a new problem arises in reproduction. What shall be the proper level for average intensity of reproduction at which a balance between bass, treble and middle register should be reached? This can be solved only by a good pair of musically trained ears listening to a diversity of records in the room or hall in question.

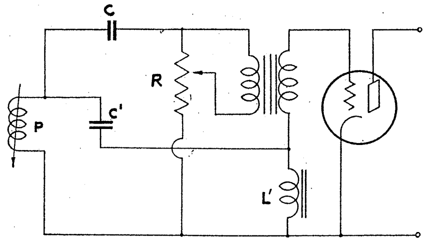


FIG. 5

Scheme for reinforcement of high frequencies.

Now that we have an idea of the problems to solve, let us see what are the inherent limitations in the records, particularly in discs as found at present in the market. In the lower register, it is apparent that if the bass tones were recorded following the same law as the middle and upper tones, the amplitude of the waves would exceed by many times the width of the groove, and it would be necessary either to make the space between grooves larger, with consequent reduction of the duration of the rendition recorded, or the outside diameter should be greatly increased. Either of these alternatives would be commercially impracticable; hence the necessity of reducing the amplitude of the low-tone waves. In the upper register, we also have a limitation which is imposed by the thickness of the needle point which increases very rapidly during the first few revolutions of the disc on account of the excessive pressure (at the start it is some thirty thousand pounds per square inch). With a thicker point, the needle cannot follow the very fine indentations in the record corresponding to the very high frequencies because they are smaller than the diameter of the section of the needle that is engaging the record groove, and if they are slightly larger the note will sound but not with full intensity. Hence, it is necessary to compensate for this defect.

So far, we have seen that we must supply something that is lacking in the record, but we must likewise eliminate other things that are not in the reproduction. Of these, two are particularly offensive and are quite common. One is the "surface noise" or needle scratch, and the other is the resonant frequencies in the whole electrical and mechanical chain that links the recorded wave with the air pressure wave that affects the ears of the listeners. In this category we find that acoustic resonance in the loudspeaker and in the surrounding space predominates.

From the preceding considerations, we note that there are four corrections to be applied to the amplifiers that furnished the necessary energy from the pickup exitation to loudspeaker operation. These are:

1. Reinforcement of the bass tones (below 100 c.p.s.)
2. Reinforcement of the treble tones (above 1000 c.p.s.)
3. Elimination of surface noise.
4. Elimination of natural periods.

Reinforcement of the Bass Tones (Below 100 C.P.S.)

The lowest frequencies found in records are confined to about 50 cycles and up. Also in radio broadcast, rarely is there anything below this mark, as it has been found almost impossible to detect the difference in quality with frequencies below 50 eliminated or with them present in the musical rendition. The pianoforte is practically devoid of foundation tone in its last octave, and in the orchestra only the bass viol is able to give a bass with strong fundamental below even 100. The bass tuba is full of overtones which are much louder than the fundamental in the 50-100 c.p.s. octave, although nominally it should go down to 43 c.p.s. For organ music, I must admit that I would be sorry to hear it deprived of the glorious 32-foot bass stops, but even then, these tones would be missed only on rare occasions as they are never alone, for the organists play them or should play them always accompanied by stops sounding one or even two octaves above, which are much more powerful. It follows that for ordinary purposes, excellent music may be obtained even with the bass end of the scale limited down to 50 c.p.s.

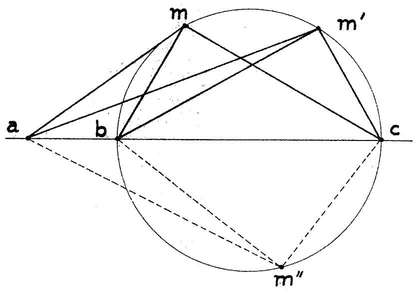


FIG. 6

Grid-filament voltage is represented by the vectors a. m.

Our aim, then, should be to reinforce the octave between 50 and 100 c.p.s. with an amplitude increasing towards the low end. There are several means of accomplishing this end, but one has been found to be extremely simple and effective, and requires but two condensers and two chokes to be added to the conventional two-stage transformer-coupled audio amplifier. The schematic circuit of one stage is shown in Fig. 1.

Let us consider the circuit comprising the pickup with its regulating potentiometer or "fader" and the condenser and inductance of the primary winding of the first transformer. If we call R the resistance (effective) of the pickup combination, or that of the plate of the tube in the case of the second stage, and if L and C are the inductance and the capacity in the series circuit, and if the resistance in the coil is negligible in comparison with the reactance, the voltage across the coil will be

$$e_2 = I\omega L \qquad (1)$$

Where I is the current and ω is the frequency in radians, the generated voltage will be

$$e_1 = I\sqrt{(R+r)^2 + \left(\omega L - \frac{1}{\omega c}\right)^2} \qquad (2)$$

the ratio of impressed to generated voltage will be

$$\frac{e_2}{e_1} = \gamma = \frac{\omega L}{\sqrt{(R+r)^2 + \left(\omega L - \frac{1}{\omega c}\right)^2}} \qquad (3)$$

and if, for simplicity, we take into account the ratios of the resistance of the tube, R, or pickup to the reactance of the coil ωL, or that of the condenser $\frac{1}{\omega C}$ at the frequency for which they are equal, and calling this ratio Q, then

$$Q = \frac{\omega L}{R+r} = \frac{1}{\omega C(R+r)} \qquad (4)$$

Calling f_o the frequency for which these reactances are equal, and designating the ratio between any frequency f_2 to f_o by n, so that $f = nf_o$, then the reactance of the coil at any frequency is $X = nX_o = nQR$, and that of the condenser will be $X_c = n^{-1}X_o = n^{-1}QR$.

The gain in voltage at any frequency between the impressed e.m.f. and the voltage across the coil (which multiplied by the ratio of transformation is the impressed voltage at the grid of the first tube) will be given by the expression:

$$\gamma = \frac{nQ}{\sqrt{1 + Q^2(n - n^{-1})^2}} \qquad (5)$$

By giving numerical values to Q, we find the corresponding values of the gain γ for various frequencies, and derived the curves in Fig. 2. The abscissae represent n, or ratios of any frequency to the resonant frequency f_o. The first curve, marked "A," was calculated with $Q=10$, while curve "B" was obtained with $Q=5$. It will be noted that the first of these two curves will not give a very satisfactory shape to the bass octave voltage curve because the gain is so sharp. If, however, we are content with a smaller gain, as in curve "B," and we follow the first stage with a second stage hav-

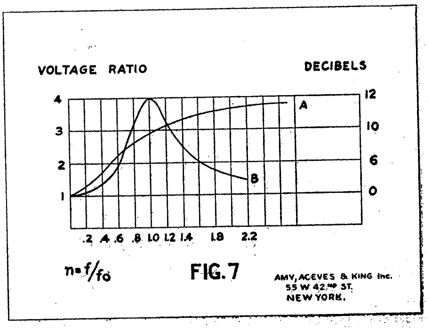


FIG. 7

Graph A indicates the relation between Eg and f.

ing a similar circuit, but the frequency f_o moved, say about 20 per cent, we will obtain curve "C" in a similar manner. If the ordinates are logarithmic, such as in our graphs, the combined gain of the two stages will be represented by the sum of the ordinates of curves "B" and "C," and this is represented by curve "D," which shows a gradual increase in level from normal (1:1 gain exclusive of tube amplification) at high frequencies to a maximum of 15 times at $n=1.7$ approximately. By making $f_o=50$ c.p.s., the bass range will be sufficiently compensated for practical purposes. If, however, the bass is too pronounced in certain records, the curves can be brought down almost flat by shunting the coils with a resistance, the values of which may vary between $R=X_o$ and $R=\infty$.

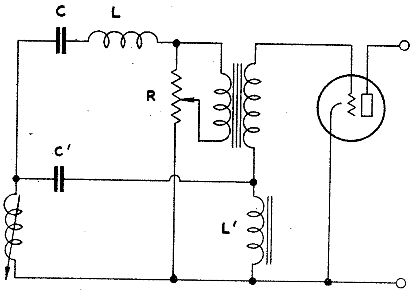


FIG. 8

Rate of change of amplitude with frequency made steeper by adding L in series with R and C.

It will be noted that it is relatively easy to change an ordinary transformer-coupled amplifier to a bass compensated amplifier. All that is required is to put a parallel feed into the plate of the tubes by means of chokes or resistances, and insert in series with the "P" terminal of the transformer primary windings, a condenser such that $4\pi^2f^2L_1C=1$, for $f=50$ in one stage and for $f=60$ in the second. It is well to observe this order, because when the first stage is tuned to $f=60$, the slightest hum will be picked up and magnified considerably. Needless to say, also, that the hum elimination becomes correspondingly more difficult when a compensated bass amplifier is fed from an a-c. source, than in the case of the straight or slightly "drooping" characteristic amplifier of the commercial types found in radio sets. In order to show the actual effects of the bass tone compensation, we will play an organ and an orchestral record through a resistance-coupled amplifier and then through the compensated amplifier. The organ selection is a Prelude and Fugue by Buxtehide played on one of the largest organs in the world at Hamburg, Germany. The orchestral number that follows is part of the immortal Fifth Symphony in C Minor of Beethoven. Another illustration is afforded by an equally great masterpiece, Tchaikovsky's Symphonie Pathetique, from which a certain passage will illustrate the dignity and grandeur of tone coming from the low register.

Reinforcement of the Treble Tones (Above 1000 C.P.S.)

The high-frequency tones are not so much reduced in relative magnitude as the bass in the average modern record. The lack of high-frequency response comes, as a rule, from worn-out needles and from stray capacities in the windings of transformers, wiring, etc. In some instances, the records themselves are lacking brilliancy, and in radio sets, an excessive selectivity may mutilate the upper band of the audio range. Some volume controls of low resistance, across a pickup, actually ruin the upper frequency response.

The evil effects begin to be felt a little above one kilocycle; hence our aim should be gradually to raise the level from about 1500 c.p.s. to about at least 3000 for good articulation, or up to 6000 for the preservation of the overtones in instrumental music, such as violin solos, where the individuality of the instrument must be kept. It is to be noted that the "apparent" loudness of a musical rendition is closely connected with the intensity of the upper register, and to see what we mean by that, let us play a record containing brass instruments doing the solo or at least in harmony, such as the second part of the "Gotterdammerung" from the opera, "Siegfried" by Richard Wagner. It will be noted that the extreme brilliancy of the overtones of the trumpets make up for the power of this musical passage. Playing it over with the overtones well subdued, the difference in power can be appreciated.

There are many ways of reinforcing the high tones, but amongst the simplest expedients are two which have several advantages:

(a) By suitable transformer design.

(b) By separate transformer to boost frequencies above a given value.

The first method (a) has fixed characteristics and hence it is not adjustable. The second (b) is adjustable and it may be given any pre-assigned shape.

In the first method (a) advantage is taken of the leakage reactance and distributed capacity in the transformer windings. Fig. 3 represents the equivalent circuit. If L_2 is about 10 times L_1, and C_2 is about the same value as C_1, which is the ordinary case in interstage transformers, we can ignore, without serious error, C_1 and L_1 and consider the circuit L^1_2 L_2 C_2 as one in which an e.m.f. is generated across L^1_2 which is common to primary and secondary (M) and the resonant circuit L_2 C_2 will allow maximum current flow when $f=\dfrac{1}{2\pi\sqrt{L_2C_2}}$ and as there is an effective resistance R^1 not shown in the circuit, it will be seen that we can reduce the circuit of Fig. 3, to the schematic circuit of Fig. 4, which is of the same type as the bass boosting arrangement previously discussed but with different characteristics. There are many transformers in the market that have a treble boosting characteristic when suitably utilized.

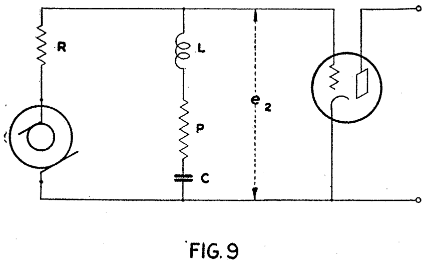


FIG. 9

Ratio of generated to impressed voltages.

The second means (b) for the reinforcement of high frequencies, which requires a minimum of additional apparatus, is shown schematically in Fig. 5. It will be noted that the impressed voltage between grid and filament of the first tube consists of two components; one which is practically constant in magnitude at all frequencies above the middle register and is in phase with the generatel voltage, and another component which varies rapidly with the frequency and comes from the circuit comprising a resistance and a condenser. The same e.m.f. is impressed across the condenser and resistance CR as it is across the choke L^1 (which may be the primary of an audio transformer) neglecting the small reactance of the condenser C_1 which, by the way, serves the purpose of reinforcing the bass as discussed in Section 1. If we make a vector diagram of these e.m.f.'s, it will be noted that the locus of the voltage across the condenser or across the resistance will move in a circle, since at point m the vectors are always at 90 degrees with respect to each other, and the impressed voltage is constant. Now, if the circle is drawn at a scale equal to the ratio of transformation the voltage from grid to filament will be represented by the vectors am. in Fig. 6. The frequency will be given by the tangent of the angle which is $\frac{1}{2}fCR$. Using a ratio of 3/1 for T, the graph that represents the relation between E_g and f, expressed in terms of $n=f/f_o$ is given in Fig. 7, Graph A.

It will be noted that the rate of change of amplitude with frequency is not sufficiently steep to accomplish the desired results. To make it steeper, an inductance introduced in series with R and C as in Fig. 8 will change the shape of curve A, Fig. 7, and will be represented by curve B, Fig. 7. There is no need of a physically separate coil; the leakage reactance of the transformer when properly chosen may accomplish similar results in practice. The control of the extent to which the treble tones are thus magnified is effected by providing the resistance R with a sliding arm like a potentiometer and connecting the transformer as in Fig. 8. It will be noted that only when the potentiometer contact includes all the resistance the vector diagrams hold true, but at other intermediate points, the steepness of the curve will be smaller. This is no serious objection, since, after all, when the arm is moved so as to reduce the effect of the treble "booster," we do not care particularly to emphasize too much a certain part of the scale since we do not need to. The steepness of the curve is determined by the ratio $Q^1 = X_o/R$ and the amount that will overamplify the treble will depend upon the ratio of transformation. In practice, a 3/1 ratio is quite sufficient, and curve C, Fig. 7 has been determined and plotted with that ratio and with Q=3.

Elimination of Surface Noises

The elimination of surface noises is a rather difficult problem because they consist mostly of excitation, impressing sharp mechanical shocks to the moving element in the pickup. Hence, any mechanical or electrical system in the whole chain which is not periodic will oscillate and that will result in surface noises. There are some other noises which come from forced vibrations due to inherent irregularities in the "grain" of the disc. These are almost impossible to eliminate unless a good portion of the upper register is sacrificed, and we have to look for further improvements in the process of manufacture of discs and in chemistry of the substances used. The natural oscillations occur at frequencies in the neighborhood of 3700 cycles in most cases. There seem to be several bands of frequencies but it has been found by trials that if the group around the first named figure is suppressed, over one-half of the total volume of surface noise is eliminated. For this reason, a tuned shunt that will reduce considerably 3700 cycles plus or minus six per cent in each side will eliminate a lot of surface noise without wiping off the entire octave from about 2500 to 5000 c.p.s. as often happens with condensers shunting the grid or plate circuits of the amplifier tubes or the output to the dynamic speaker. The elimination of only three notes is hardly noticeable because these high frequencies occur practically in all instances as overtones and very rarely as solo tones. Therefore the color of the tone of a few notes will suffer to some extent, while the neighboring notes contain all their overtones, and it will be quite difficult even to a trained musician to distinguish when these three frequencies are present or when they are suppressed in a rendition even of the best of the classics.

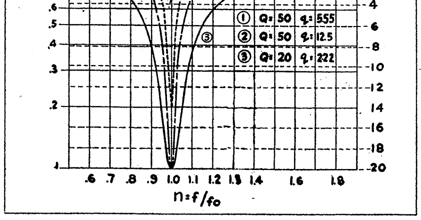


FIG. 10

The ordinates represent voltage ratios; the abscssae, frequency ratios.

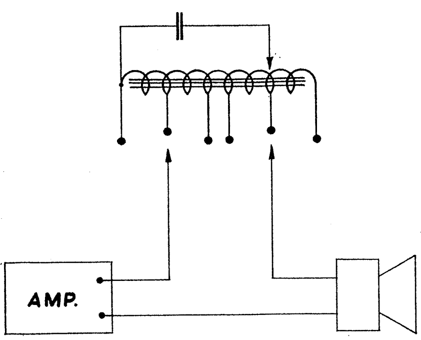


FIG. 11-A

Band elimination by series tuned traps.

The result of a tuned shunt is well-known in its mathematics. We will point out only a way in which the design of the coil and condenser may be readily obtained. If we call R the resistance of the source (tube or pickup), Fig. 9, the resistance P of the resonating shunt (of which practically all is in the coil) X_o the reactance of the coil for a frequency f_o, such that $4 \pi {}^2 f^2_o LC = 1$, and calling $f/f_o = n$, as before, the ratio of the generated to impressed voltages e_1 and e_2, Fig. 9, will be

$$\gamma = \frac{e_2}{e_1} = \frac{\sqrt{R^2 + (\omega L - \omega^{-1} C^{-1})^2}}{\sqrt{(R + P)^2 + (\omega L \times \omega^{-1} C^{-1})}}$$

and in terms of the ratios of initial reactance X_o to resistances which we shall call $Q = X_o/R$ and $q = X_o/r$

$$\gamma = \sqrt{\frac{Q^{-2} + (n - n^{-1})^2}{Q^{-1} + 9^{-1})^2 + (n - n^{-1})^2}}$$

By giving numerical values to Q and q, we have obtained the curves in Fig. 10. It will be seen that the shape is controlled mostly by Q and the reducing effect by q, although both of them contribute to the sharpness as well as to the magnitude of the effect. In this graph, the ordinates represent the voltage ratio γ in decibels below normal level, and the abscissae are in terms of frequency ratios, f/f_o. By making f_o about 3700 c.p.s. the best results will be obtained for the suppression of surface noises with the pickups that were used in the experiments, but it is safe to say that for most pickups the frequency under consideration will not differ materially. It is possible that for best results one or more tuned shunts should be employed, but in the majority of instances one is quite sufficient.

The elimination of single frequencies, or narrow bands, can be accomplished as well by series tuned traps, Fig. 11, or by combinations of series and shunt circuits forming T or sections. The calculations for a series rejector are too well-known to insist upon them, but unless such circuits are used where current is drawn and not merely potential, as when used between tubes, they will accomplish very little or nothing. The best place for series traps is either between the pickup and input network or between the output of the amplifier and the speaker. If the inductor is provided with taps, the effect may be increased or diminished by changing the tap without affecting the tuning, as shown in Fig. 11-A.

Elimination of Natural Periods

The suppression of resonating frequencies, whether due mostly to defects in the speaker or to poor acoustic conditions, presents a very similar problem to that of the elimination of surface noises. In the case of the scratch noise, the frequencies should be practically wiped out, whereas in the case of the resonance problem they should only be reduced in magnitude to such an extent that they will come at the same level as the neighboring frequencies. In shape, the acoustic resonance graph is similar to the tuning characteristics of radio sets, and although not so sharp as the combined effect of several r-f. stages, it may be sharper than a single stage of tuned r-f.

The calculations for the design of a tuned shunt are exactly the same as in the case of the scratch elimination shunts; the proper proportions of q and Q should be chosen. However, for a given shunt which may take care of the most severe cases, (and by severe is meant sharp resonance), the effect can be at the same time subdued as well as broadened by the insertion of series resistance. It will be noted that the magnitude and the sharpness go hand in hand in the case of the acoustic resonance, for the sharper the resonance of the mechanical system, whether coming from the properties of the loudspeaker alone or from resonating spaces near it or in the room itself, the greater the maximum amplitude or resonance will be. So it is with the tuned shunt; the smaller the resistance within the shunt, the greater the short circuiting effect that it will possess for the selected frequency, and the smaller the effect (relatively speaking) on the neighboring frequencies.

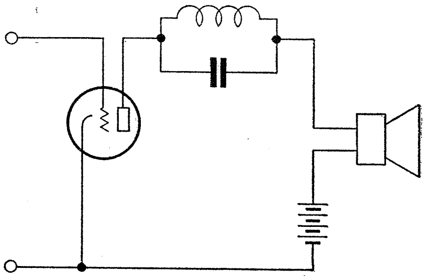


FIG. 11

Inductor provided with taps.

Up to the present we have seen how an existing amplifier may be given the desired characteristics to compensate

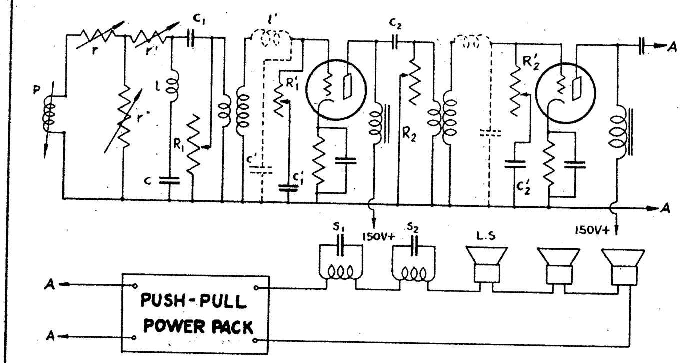


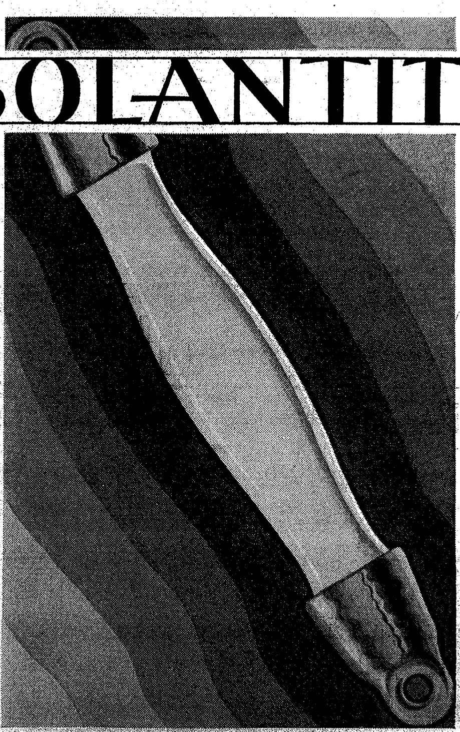
FIG. 12

Fig. 12. 15 watt amplifier to operate from pickup.

for the deficiencies in the processes of recording and reproducing. It is well to show how easy it is to build an amplifier containing all the features that we have described, and which will have sufficient gain for ordinary purposes. Fig. 12 represents an amplifier designed to operate from a phonograph pickup to an output of about 15 watts. The phonograph pickup P delivers its voltage to a constant resistance "fader" made of three resistances, r, r' and r", and immediately a scratch shunt 1c follows. The first stage contains a transformer with series condenser C_1 and shunt resistance R_1 for the purpose of controlling the bass reinforcement and regulating the amount of this gain. The dotted lines in the secondary circuit represent the leakage inductance and the distributed capacity of the secondary, which by suitable transformer design may be made to reinforce the upper register. To control the gain in this part of the scale, a resistance R_1^1 with a condenser C^1_1 in series with it may shunt the secondary.

The second stage contains substantially the same apparatus but the frequencies to which the capacities and inductances are adjusted should be about 10 to 15 per cent different from the values used in the first stage for reasons discussed in section 1. The push-pull stage contains only a series condenser which will reinforce the bass. It is left without control because all records are deficient in bass and therefore there is no need of reducing the little gain which it brings in.

The resonance of the natural periods of the speakers or of the auditorium are corrected by means of shunt traps S_1 S_2 in the output circuit of the amplifier where they belong, since these natural periods vary in different localities and they may be totally absent, at least appreciably, in many places. It will be noted that they can be adjusted in frequency by varying the number of turns included in the condenser circuit, and in attenuation factor by varying the number of turns included in the connection to the line. The tubes are of the separately heated cathode type, the heater circuits of which are not shown. The four section electrolytic condenser with two 8 and two 18 microfarad sections is of standard make and will take care of the low tension filtering and by-passing. The chokes and resistances will act as sufficient filtering series elements, even in the case of high gain 60 cycle amplification. Care must be exercised to avoid magnetic interlinkage between interstage transformers and the power transformer and choke, as a very small amount of induction will be greatly magnified. The performance of an amplifier such as has been described and illustrated is similar in nature whether the audio frequencies come from a pickup, a photocell or the detector tube of a radio set. In commercial applications, the chokes used to feed the plates of the amplifiers may be substituted by resistances, so long as a high voltage source is available, which is the most general case even when —45 type tubes are placed in the power stage. By suitable selection of thickness of laminations and arrangement of the windings of the interstage transformers, the controlling resistances R_1, R^1_1, R_2, R^1_2 may be eliminated, but the performance of the amplifier will be fixed and not adjustable, which, by taking average conditions, may be acceptable in less expensive amplifiers.



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CONTRIBUTORS to the Proceedings, by bearing in mind the points below, will avoid delay and needless expense to the Club.

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2. Illustrations should invariably be in black ink on white paper or tracing cloth. Blueprints are inacceptable.

3. Corrected galley proofs should be returned within 12 hours to the office of publication. Additions or major corrections cannot be made in an article at this time.

4. A brief summary of the paper, embodying the major conclusions, is desirable.

5. The Club reserves the right of decision on the publication of any paper which may be read before the Club.

* For 1930 the Chairman of the Papers Committee is Mr. F. X. Rettenmeyer, 463 West Street, New York City.