Line Filter for Program System *

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Open wire circuits recently have been developed for transmitting radio broadcast programs with greater naturalness and over greater distances than heretofore.[1] The simultaneous utilization of these circuits for the transmission of broadcast programs and carrier telephone messages requires the use of line filters to restrict the program and carrier currents to the proper circuits. The low pass line filter developed for the program circuits and its contribution to the maintenance of good quality in the programs transmitted are described in this paper.

PROGRAM transmission systems operated on open wire telephone lines ordinarily are not assigned the exclusive use of the lines, but usually share them with other communication facilities. The wide-band system described in an accompanying paper [1] transmits currents in the frequency band extending from 35 to 8,000 cycles per second, while the lines over which it is routed possess useful transmission ranges extending from 35 to considerably above 30,000 cycles. In order that the range above 8,000 cycles shall not be wasted, carrier telephone systems utilizing these frequencies usually are operated on the same wires with the program systems.

Line filters are used at each terminal and repeater point in the program system to separate the program currents from the carrier currents and to guide each to the proper channel. They are operated in sets consisting of a low-pass filter and a high-pass filter connected in parallel at one end, the end that faces the line. The low-pass filter transmits the program currents freely while effectively excluding the carrier currents, and the high-pass filter transmits the carrier currents while excluding the program currents.[2]

The line filters are located in the open-wire program systems as shown in Figs. 1, 15, 16, 17, 18, and 20 of the accompanying paper by H. S. Hamilton.[1] The low-pass filter is in the direct path of the program currents and therefore has a number of features of special interest. It is the object of this paper to describe this filter and its contribution to the maintenance of good quality in the programs transmitted over the system.

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[1] "Wide-Band Open-Wire Program System" by H. S. Hamilton, published in this issue of the *Bell. Sys. Tech. Jour.*

[2] "Telephone Transmission Networks" by T. E. Shea and C. E. Lane, published in *A. I. E. E. Transactions*, Vol. 48, 1929, pages 1031–1034.

This low-pass line filter, with its associated high-pass filter, makes it possible to use the open-wire lines simultaneously for wide-band program service and for commercial carrier telephone service, without impairing the quality of the program. It represents an improvement over older types of line filters, as well as an advance in the technique of equalization in filters. In cases requiring careful delay and loss equalization, it has been the usual practice to design the filter first to supply the required discrimination or filtering action, and then design a delay corrector to correct for the delay distortion in the filter, after which a loss equalizer is designed to correct for the amplitude distortion in both the filter and the delay corrector. The loss equalizer introduces a small delay distortion which usually can be anticipated and corrected in the delay corrector. In the wide-band program filter the functions usually performed by these three separate types of networks have been combined, with a consequent saving in cost and space.

Requirements to Be Met by Program Filter

To function effectively as a line filter, the low-pass filter must provide sufficient discrimination against carrier currents to make their effect completely inaudible in all the receivers connected to the program system. Discrimination varying from 46 to about 90 db is necessary to accomplish this end. Because of the presence of an auxiliary low-pass filter [1] which supplies considerable loss in the frequency ranges where the requirement is unusually severe, each line filter need furnish discrimination varying only from 40 to about 60 db.

From the standpoint of program quality, it is essential that the line filter, while furnishing the foregoing discrimination, shall not introduce any appreciable distortion into the program. This requirement would call for nothing unusual in the way of filter design if there were only a few filters in the system. Long open-wire program systems, however, may extend as far as 3,000 or 4,000 miles, and may contain as many as 50 low-pass line filters. A program that has traversed such a circuit still must be comparable in quality to a program that is broadcast from the point at which it originated. Since the system contains much other apparatus, such as equalizers and amplifiers, each low-pass line filter can be permitted to introduce not more than about 1/100 of the distortion that can be tolerated in the whole system, assuming 50 filters in the system.

There are two types of distortion that must be controlled very carefully in the program filter: these are (1) amplitude distortion, and (2) delay, or phase, distortion. Amplitude distortion is introduced by a filter when its loss is not the same at all frequencies in the transmitted band, currents of some frequencies being attenuated more than others. The effect of amplitude distortion on the program is to change the relative intensities, or volumes, of tones of the frequencies at which distortion occurs, thus impairing the naturalness of the program. Amplitude distortion ordinarily can be corrected without much difficulty by means of suitable attenuation equalizers.

Delay distortion is introduced by a filter when different frequency components of a signal require different lengths of time for propagation through the filter. This type of distortion is related directly to the shape of the phase shift-frequency characteristic. The slope of this phase shift curve usually is taken as a measure of the delay introduced by the filter. Stated mathematically, the delay in seconds is taken as $\partial B/\partial \omega$, where B is the phase shift in radians and ω is $2\pi f$, f being the frequency in cycles per second. Thus if the phase shift of the filter is proportional to frequency, $\partial B/\partial \omega$, or the delay, is constant and there is no delay distortion. In this case the wave form of a signal transmitted through the filter remains unchanged, the signal being delayed in transmission an interval of time corresponding to the slope of the phase shift curve. If the slope of this curve is not constant over the transmitting band of the filter, however, delay distortion is introduced. In low-pass filters, the difference between the slope of the phase shift curve at a given frequency and the minimum slope of the curve is a measure of the delay distortion at that frequency.

A discussion of delay distortion in telephone apparatus, including filters, as well as a discussion of the effect of delay distortion on telephone quality, may be found in two recent articles on these subjects.[3,4] Whereas the effect of amplitude distortion is to weaken or strengthen some of the tones in the sound being transmitted with respect to the other component tones, the effect of delay distortion is to introduce unnatural audible effects which may become so pronounced as to be annoying if the delay distortion be great enough.

Delay distortion is present in most filters used in communication work, but ordinarily not in such magnitude that its effect is noticeable. As a rule, it need be considered only when a large number of filters is used in a single circuit, as in the case of the program systems. Delay distortion is in general more difficult to correct than amplitude distortion. One of the unusual features of the low-pass line filter used in the wide-band program circuits is the means employed to keep it free from delay distortion.

[3] "Phase Distortion in Telephone Apparatus" by C. E. Lane, *Bell. Sys. Tech. Jour.*, July, 1930.

[4] "Effects of Phase Distortion on Telephone Quality," by J. C. Steinberg, *Bell Sys. Tech. Jour.*, July, 1930.

The filter consists of four parts, each with distinguishing functional characteristics. The separate parts, or sections, have image impedances such that when they are joined together no current is reflected at the junctions. Figure 1 shows the filter in block schematic form.

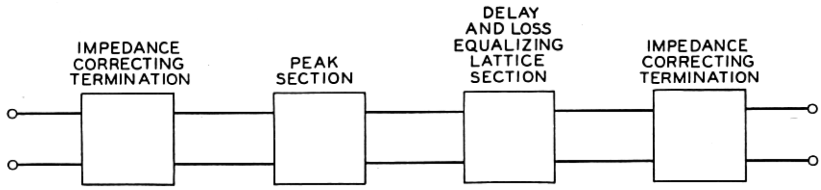

Fig. 1—Block schematic diagram of filter.

Each part of the filter provides some of the attenuation required to exclude carrier currents from the program circuit, the attenuation of the complete filter being the sum of the attenuations of all parts. On Fig. 2 are shown the loss-frequency characteristics of the various sections and of the complete filter.

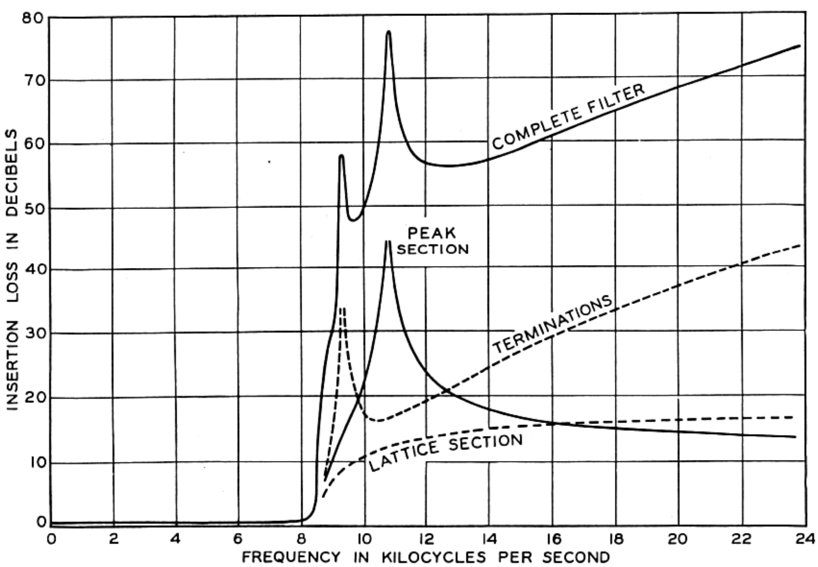

Fig. 2—Loss in filter and in component sections.

Delay Equalization

Likewise, the phase shift of the complete filter is the algebraic sum of the phase shifts of all sections. The phase shift of the filter exclusive of the delay and loss equalizing section is similar to that of the usual ladder type low-pass filter. Over the lower frequencies of the transmitting band the phase shift-frequency characteristic is practically linear with frequency, but at the higher frequencies the slope of this curve increases gradually with frequency and becomes very large near the upper edge of the band. Phase shift varying in this manner introduces much more delay distortion than can be tolerated, and therefore has to be corrected. It is one of the functions of the delay and loss equalizing section, which is of the lattice type, to correct for this distortion. The phase shift of this lattice section is such that when it is added to that of the rest of the filter the total phase shift is very nearly proportional to frequency over the whole program band, and delay distortion thus is almost entirely eliminated.

The property of the lattice section by which its phase shift can be made to vary with frequency in the desired manner is expressed in the following characteristic equation, which holds only in the transmitting band and when the section is terminated in its image impedances:[5]

$$\tan\frac{B}{2} = \frac{Kf\left(1 - \frac{f^2}{f_2^2}\right)\sqrt{1 - \frac{f^2}{f_c^2}}}{\left(1 - \frac{f^2}{f_1^2}\right)\left(1 - \frac{f^2}{f_3^2}\right)}. \tag{1}$$

In this equation, B is the phase shift in radians; f is the frequency in cycles per second; f_1, f_2, f_3, and f_c are frequencies at which the phase shift of the section is successive multiples of π radians or 180 deg., f_c being also the cut-off frequency of the filter; and K is a constant controllable by assigning the proper values to the coils and condensers of the section. By assigning to f_1, f_2, and f_3 the values of frequency at which it is desired that the phase shift of the section shall be π, 2π, and 3π radians, respectively, and by giving K the proper value, the phase shift-frequency curve is made to approximate the ideal one which completely would correct the delay distortion of the filter. Figure 3 illustrates the building up of the phase shift characteristic.

The delay corresponding to the rate of change of the phase shift with frequency is plotted in Fig. 4. The average delay introduced by the filter is about 0.00035 sec. It may be noted that for frequencies below 7,500 cycles per second, the variation from this average does not exceed 0.000025 sec. Thus the delay due to 50 filters in a long program circuit does not deviate from the average in this frequency range by more than 0.00125 sec. Distortion of this amount ordinarily would not be detected by the average listener. Above 7,500 cycles

[5] U. S. Patent No. 1,828,454 to H. W. Bode.

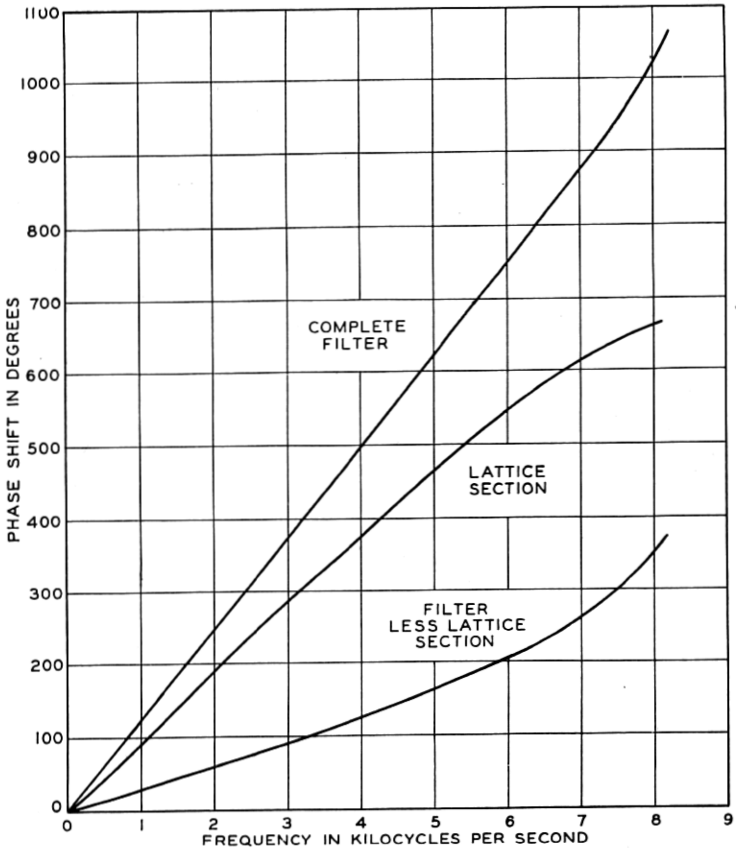

Fig. 3—Phase shift in filter and in component parts.

per second the delay gradually increases with frequency, rising quite rapidly outside the program band. The high attenuation at frequencies above the program range, however, eliminates any effect this distortion otherwise might have on the program.

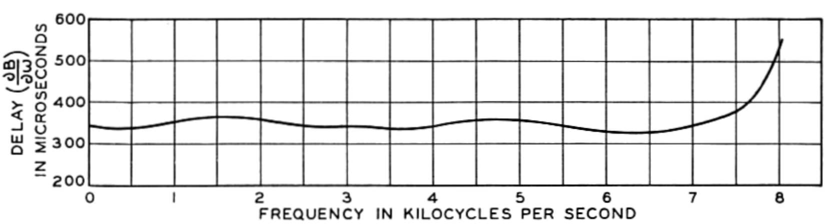

Fig. 4—Delay-frequency characteristic of filter. The ordinates of this curve are proportional to the slope of the upper curve of Fig. 3.

Loss Equalization

Another function of the lattice section is to make the loss of the filter constant in the program frequency band. In a dissipationless filter terminated in its image impedances (which is substantially the condition under which this filter is operated) the loss in the transmitting band is zero. The effect of dissipation is to introduce a loss which is given approximately in this band by the equation:

$$A_d = \frac{\omega}{2Q}\frac{\partial B}{\partial \omega} \tag{2}$$

where A_d is the loss due to dissipation, B is the phase shift of the non-dissipative filter, and Q is the average dissipation factor of the coils (dissipation in the condensers being negligible, ordinarily). The factor Q is equal to the average of the ratios $\omega L_e/R_e$, and $\omega/2Q$ in equation (2) therefore may be written $R_e/2L_e$, where R_e and L_e are the effective resistance and effective inductance, respectively, of the coils.

In the coils of the program filter, Q is about proportional to frequency over the lower portion of the program band, but above this range the factor $\omega/2Q$ increases with frequency. For the filter exclusive of the lattice section, the factor $\partial B/\partial\omega$ is also greatest at the higher frequencies, as may be seen from the lower curve in Fig. 3; hence this part of the filter introduces much more amplitude distortion than is permissible. For the lattice section alone, however, the factor $\partial B/\partial\omega$ is greatest at the lower frequencies, as is apparent from the middle curve of Fig. 3. Thus the natural tendency of dissipation in the lattice section is to compensate for the distortion in the other sections of the filter. This compensating tendency can be controlled to a considerable degree, since by equation (2) A_d is proportional to R_e. By proper adjustment of the effective resistance of the coils of the lattice section, its loss is made practically complementary to that of the rest of the filter, so that the loss of the complete filter is substantially constant throughout the program range.

The loss of the filter in the transmitting frequency band is shown in Fig. 5. The average loss below 7,000 cycles per second is about 0.53 db and the deviation from this average does not exceed 0.03 db. Considering again a circuit containing 50 filters, the deviation from the average loss introduced by the filters does not exceed 1.5 db in this range. Between 7,000 and 7,500 cycles per second the amplitude distortion per filter is about 0.10 db, and above 7,500 cycles the loss increases in such a way as to tend to mask the small delay distortion in this range.

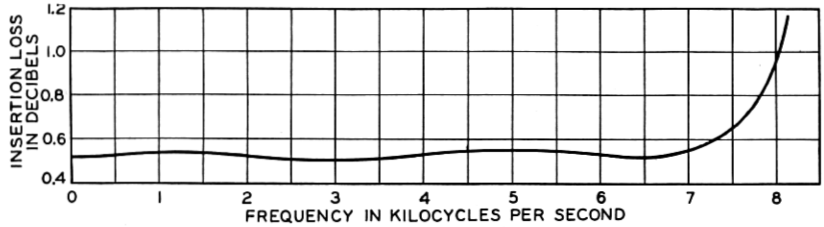

Fig. 5—Loss of filter in program frequency band.

Impedance Correction

In the discussion of the lattice section it was stated that its phase shift is given by equation (1) only when the section is terminated in its image impedance. To facilitate the design and simplify the filter structure, this section has been given an image impedance of the simplest type. This impedance, Z_I, varies with frequency according to the following equation:

$$Z_I = \frac{Z_o}{\sqrt{1 - \frac{f^2}{f_c^2}}}, \tag{3}$$

where Z_o is the "nominal impedance" of the filter, a constant equal approximately to the average impedance of the open-wire lines in the program band; and f_c is the theoretical cut-off frequency. Thus the image impedance rises with increasing frequency to a very high value near the cut-off; and, since the line impedance is practically constant except at very low frequencies, a large mismatch would result at the upper edge of the transmitted band if the lattice section were connected directly to the line. The impedance correcting sections at the ends of the filter are employed to avoid this mismatch. The properties of these sections are such that when they are inserted between the lattice section and the line or the office terminating apparatus, the impedance of the filter matches that of the line and the office apparatus, and the lattice section faces its own image impedance. In this manner, both internal and external reflections largely are avoided; and the phase shift of the lattice section has the proper value.[6]

The general theory on which the design of the impedance correcting sections is based is discussed at length in a recently published article.[7] In brief, the sections consist of two parts: a 4-terminal network to

[6] "Impedance Correction of Wave Filters," by E. B. Payne, *Bell. Sys. Tech. Jour.*, October, 1930.

[7] "A Method of Impedance Correction," by H. W. Bode, *Bell. Sys. Tech. Jour.*, October, 1930.

make the resistance of the filter approximately constant over the program band, and a 2-terminal network placed in shunt at the end to cancel the reactance of the filter in this band. The inductance and capacitance of the coils and condensers of the 4-terminal network are related to the coefficients of a power series expansion of the right-hand part of equation (3) in the manner explained in the article by H. W. Bode.[7] The 2-terminal shunt network at the apparatus end is designed so that, while canceling the reactance of the filter in the program band, it resonates just above the band to produce a peak or sharp maximum of attenuation. It thus supplies the sharp selectivity required to produce an abrupt change from free transmission of the program frequencies to high attenuation of the carrier frequencies.

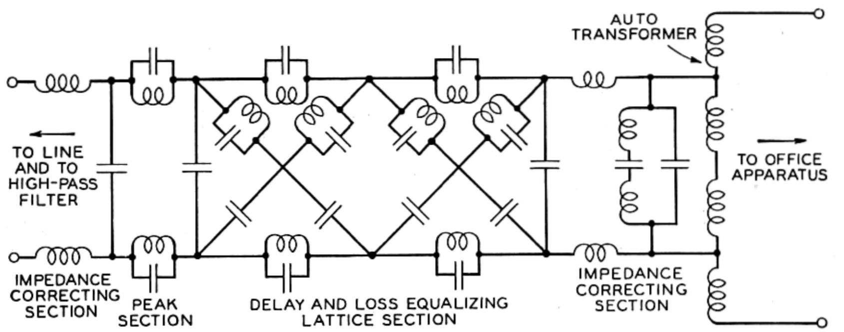

Fig. 6—Schematic diagram of filter.

At the line end, the impedance correcting section is designed for parallel connection with the high-pass line filter. The high-pass filter itself acts as the shunt reactance-canceling network.

The peak section shown at the left of the delay and loss equalizing section in Fig. 1 provides attenuation which rises rapidly with frequency above the program band in such a way as to add to the selectivity of the filter. It is a ladder section of a type often employed in filters for its selectivity.

The filter is designed to match the average impedance of the open-wire lines. The impedance of the office apparatus, however, is slightly higher than that of the lines and the filter. An autotransformer therefore is used at the end of the filter connected to the office apparatus, to effect the required change in impedance. A schematic diagram of the complete filter is shown on Fig. 6, the parts being marked for identification in accordance with the foregoing discussion.