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J. C. STEINBERG

1,836,824

WAVE TRANSMISSION WITH NARROWED BANDS

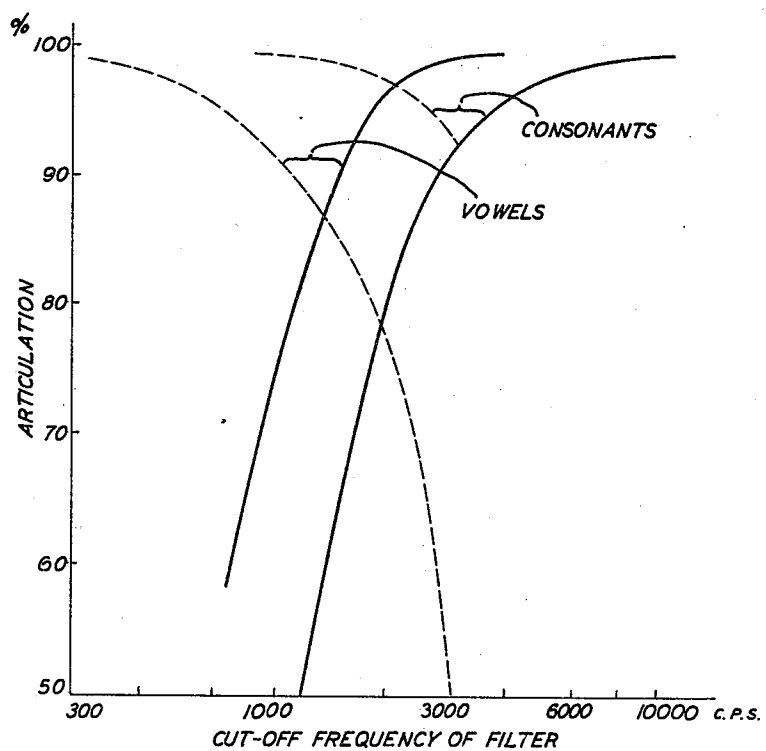
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3 Sheets-Sheet 1

FIG. 1



FIG. 2



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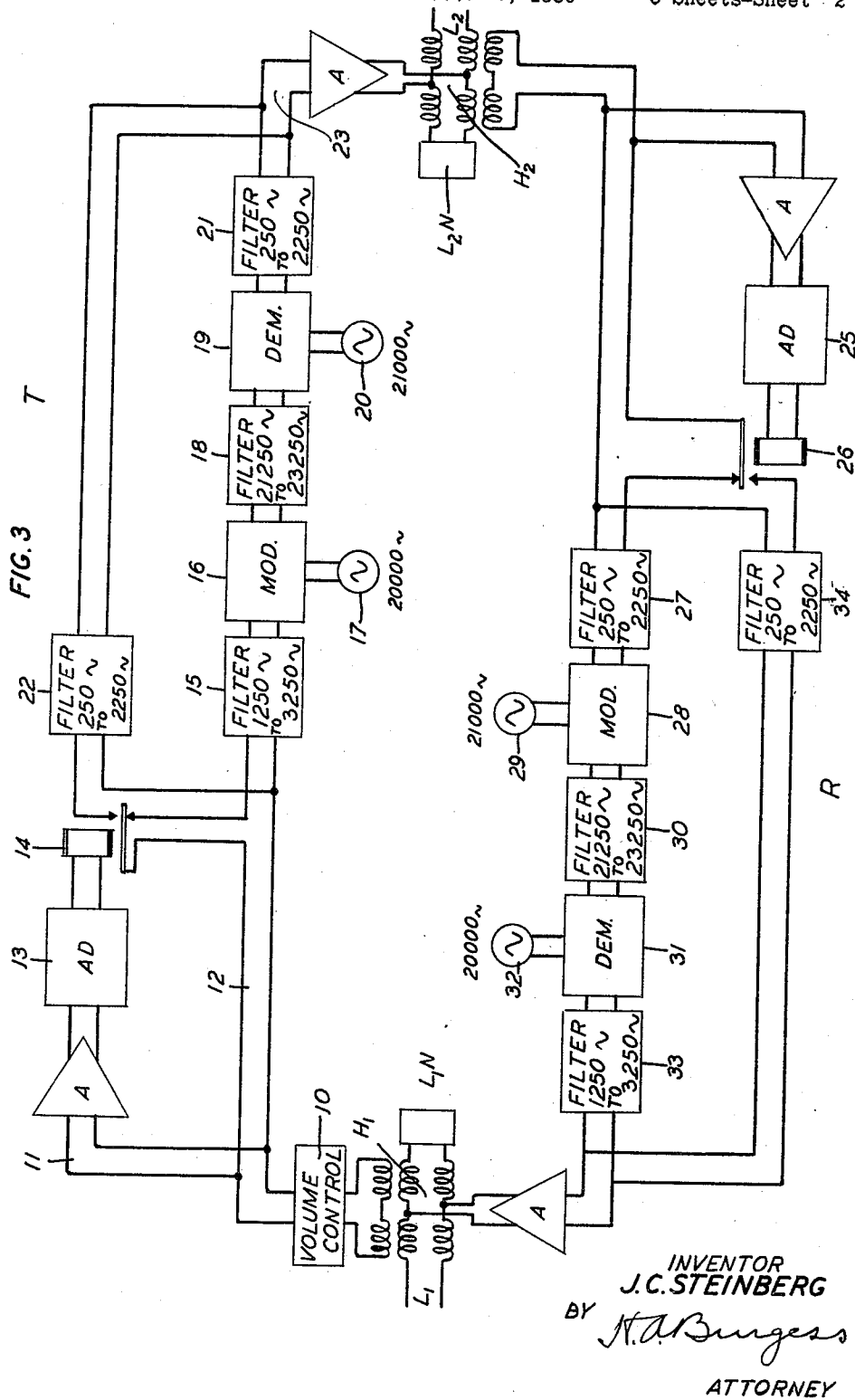
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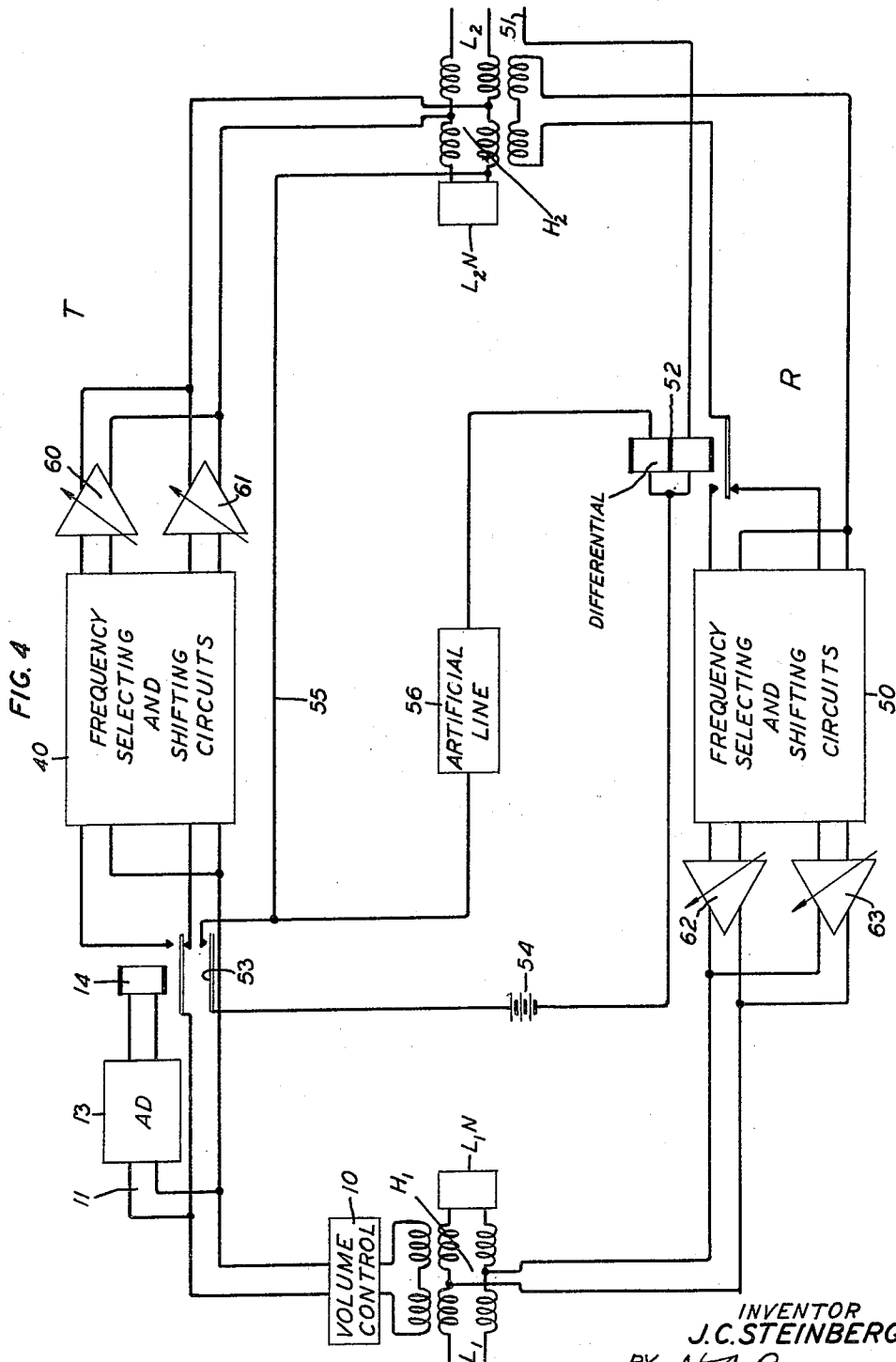
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WAVE TRANSMISSION WITH NARROWED BANDS

Application filed October 18, 1930. Serial No. 429,539.

The present invention relates to a method and system for transmitting intelligence in which the total frequency range required is less than that embraced in the intelligence-conveying waves as produced.

The invention will be described with particular reference to speech waves, although the invention is equally applicable to other kinds of waves such as music as well as waves used to produce visual effects or the like.

The invention depends upon the existence of certain statistical characteristics in the waves, which enable their subdivision into components of different amplitude and frequency level, varying from instant to instant. The components occurring at any one small instant of that time do not fill the entire frequency range occupied by the waves occurring over a considerable time. Advantage of this fact is taken by shifting the frequency level of certain components where they exist substantially alone at any instant so that they occupy the same frequency level as (or at least overlap in frequency) the components of other frequencies existing substantially alone at some other instant of time.

In the case of speech, it has been determined that the lower portion of the total frequency range is predominantly occupied by the vowel and semi-vowel sounds. The upper portion is occupied principally by the stop and fricative consonant sounds. Moreover, both classes of sounds do not exist in speech in the same instant of time but they occur in rapid succession. This fact makes it possible to divide the one class of sounds from the other on a time basis. Also the amplitudes of the vowel and semi-vowel sounds are larger than those of the stop and fricative consonants. The two classes can, therefore, be distinguished from each other in time, in frequency range and in amplitude or energy.

In practicing the invention, filters may be used to separate the low frequency—large energy components from the high frequency—low energy components. A device responding differently to the energies represented by these two classes of components, switches one or other of the filters into circuit. One filter passes the components directly to the line. The other leads to a fre-

quency-shifting circuit such as a modulator which changes the frequency level of these components so that they occupy the same range as those passed by the other filter, or at least overlap that range. Thus both classes of sounds are transmitted in a reduced band width.

At the receiver a second device differently responsive to the amplitude of the two classes of wave components controls connection of the line to a pair of circuits one of which leads directly to the receiver and the other to a frequency shifting circuit which effects a change opposite to that produced in these components at the transmitter.

Instead of using a marginally operating device at a receiver for separating the two classes of components, the receiver may be synchronized with the transmitter by means of a pilot channel. In this case, the gain of the channel transmitting the less intense sounds may be increased at the transmitter relative to the gain of the channel transmitting the more intense sounds. This will better enable the weaker sound components to override noise on the line and will give a better signal-to-noise ratio. The gain that is introduced in the branch transmitting the less intense sounds may be adjusted to suit conditions. For example, it may be made to depend upon the character of the noise on the line or it may be sufficient to compensate for unequal attenuation by the system of the different frequency components transmitted over the line. In some cases it may be desirable to introduce complementary losses at the receiver in order to restore the speech components to their natural relative level.

While a division into two classes of wave components has been referred to for simplicity, it is to be understood that the same principle can be carried farther and a large number of subdivided classes of components can be similarly employed.

The invention will be more fully set forth in the following detailed description of certain illustrative embodiments showing the preferred manner of practicing it, and the claims will define the invention not only as embodied in these illustrative examples, but also in a scope to embrace various other

forms which it is capable of assuming in practice.

In the accompanying drawings, Figs. 1 and 2 show graphs that will be referred to in describing the nature of the invention, and Figs. 3 and 4 show terminal circuit diagrams, in schematic form, of systems for carrying out the inventive idea.

The three characteristics of speech sound upon which the practice of the present invention depends, namely frequency, amplitude and time of occurrence, are graphically illustrated in Fig. 1 which shows an actual oscillogram taken of the word "seems". For convenience of showing, this oscillogram has been condensed so that relatively few undulations corresponding to each sound are given. In actual practice and in typical speech, the length of each sound would be considerably greater than shown in this figure. This oscillogram shows, however, that the "e" and "m" sounds are of much greater amplitude than the sibilant sound "s". Also, the frequency of the sound "s" is illustrated by this oscillogram as being composed entirely of higher frequencies. It would be impossible to tell for certain from this oscillogram whether the "e" and "m" sounds might possess higher harmonics, but at least the fundamentals are seen to be of much lower frequency. It is also quite obvious that the sounds occur in different instants of time.

Considering now the difference in amplitude between the vowel and semi-vowel sounds on the one hand and consonant sounds on the other, the following table has been prepared from articulation tests based upon a large number of observations in which differences in strength of the different speakers' voices and other possible sources of error have been compensated for. In this table the peak powers are given in decibels (db.) above the sound of least intensity i. e. (th).

Vowel and semi-vowel sounds	Average peak power (db.)	Consonant sounds	Average peak power (db.)
r	30.1	sh	20.4
a	28.4	ch	17.2
50 a'	28.1	zh	16.0
ā	27.2	s	14.8
e	26.9	z	14.8
u	26.7	v	13.9
o	26.5	j	13.8
55 ō	26.3	t	12.0
i	25.4	t'	9.5
ē	24.9	g	9.0
ū	23.7	b	8.5
w	----	p	7.8
60 y	----	k	7.8
l	21.1	d	6.0
m	20.4	f	4.8
ng	19.8	th	0
n	16.7	h	----
65 Average	24.8	Average	11.0

From this statistical study of speech, it is seen that the vowel and semi-vowel sounds as a class are of the order of 10 db. or more greater intensity than the consonant sounds.

Similar articulation tests have resulted in the curve of Fig. 2 showing the relative frequency ranges occupied by the vowel and semi-vowel sounds on the one hand and consonant sounds on the other. In making these curves, variable cut-off filters were used to explore the entire speech frequency range and the articulation corresponding to each frequency of cut-off was plotted. The solid line curves are those obtained by the use of low pass filters of variable cut-off, while the dotted line curves were obtained with high pass filters. These curves show that nearly perfect articulation of the vowel sounds is obtained in a frequency range from about 250 cycles per second to about 2300 cycles per second and that the improvement in extending the range in either direction beyond these limits is practically insignificant. These curves also show that for substantially perfect articulation of the consonant sounds, the range from 1000 to 10000 cycles must be used, but for a practical degree of quality, a much narrower range is tolerated. For example, the dotted curve shows that not much in articulation is gained by extending the range below about 1200 cycles per second, and the solid curve is extending upward quite gradually in the region of 4000 cycles per second. These curves will be of assistance in designing a system in accordance with the invention to meet particular conditions in practice. The actual frequency range for the consonants and also that for the vowels to be employed in any particular case will depend upon the requirements to be met and no set limits can be given.

The application of the invention to a particular system will first be described in connection with Fig. 3 which shows a two-way terminal circuit for interconnecting the subscriber's line L_1 at the left with the main telephone line L_2 shown at the right.

It is assumed in this illustrative example that the maximum frequency which it is practicable to transmit over the line L_2 is about 2250 cycles. Of course, speech could be transmitted over this line employing the range from about 250 cycles to 2250 cycles and commercial quality for many purposes would be obtained. However, in case it is desired to obtain a higher quality of transmission than would be possible with the range 250 to 2250 cycles by direct transmission, the method of the present invention permitting utilization of the range 250 to 3250 cycles per second will be described showing how an increase in quality for this available total line range can be obtained.

The subscriber's line L_1 and the telephone line L_2 are each shown as provided with the usual hybrid coils H_1 and H_2 respectively, and the usual line balancing networks L_1N and L_2N respectively.

For transmitting from the line L_1 to the line L_2 , there is provided a path generally indicated by T and for transmitting in the opposite direction between L_2 and L_1 there is provided a second path generally indicated by R.

In the transmitting path T beginning at the hybrid coil H_1 there is first a volume control circuit for reducing to the same level speech waves received over different length subscribers' lines or from individuals having different voice strength. This volume control may be of any suitable type, either automatic or manual, but preferably automatic, examples of this type being given in patent applications of R. C. Mathes Serial No. 372,951 filed June 22, 1929, and D. Mitchell Serial No. 329,203 filed December 29, 1928. This volume control when used in the present system will have a relatively large time lag so that its adjustment will not be changed for instantaneous variations in speech level but so that it will regulate only the average speech level over a considerable period of time.

On the output side of the volume control 10 the circuit branches into the two portions 11 and 12. Branch 11 contains suitable amplification and leads to an amplifier detector circuit 13 which in turn controls relay 14. This amplifier detector circuit will in practice generally by a vacuum tube amplifying and rectifying circuit for converting speech waves in the circuit 11 into direct current suitable for actuating relay 14. Suitable bias will be used on the grids of the vacuum tube 13 or suitable mechanical or electrical bias on relay 14 so that this relay is marginal and does not respond to all currents, but only to currents corresponding in strength to the vowel or semi-vowel sounds.

Circuit 12 leads directly to filter 15, through the normally closed back contact of relay 14. This filter 15 is in the consonant branch, which branch is closed unless relay 14 is actuated by vowel sounds. Filter 15 is designed to pass currents of frequencies between 1250 cycles per second and 3250 cycles per second but to suppress currents of both lower and higher frequencies. This and the other filters throughout the system may be designed in accordance with the principles laid down in the U. S. patent to G. A. Campbell 1,227,113, patented May 22, 1917.

The output side of filter 15 is connected to modulator 16 which is preferably of the type disclosed in U. S. patent to Carson 1,343,306, dated June 15, 1920. This modulator is supplied with carrier waves from source 17, these waves having a frequency of 20000 cycles per second. This source is preferably a vacuum

tube oscillator such as is commonly used in carrier wave systems. As explained in the Carson patent, this type of modulator prevents the unmodulated carrier components from the source 17 from passing into the outgoing circuit so that only the two side bands resulting from the modulating action of the circuit appear in the final output.

The output of modulator 16 is connected through a second filter 18 which is designed to pass only the range from 21250 to 23250 cycles per second, this being the upper side band resulting from the modulation in circuit 16.

This side band is then demodulated in circuit 19 by means of a sustained wave of 21000 cycle frequency generated at source 20. This demodulator 19 may be identical with the modulator circuit 16 and is of the type disclosed in the Carson patent referred to. As the result of the demodulation in circuit 19, a lower side band extending from 250 cycles to 2250 cycles per second is produced and this passes through the filter 21, the other components of modulation being outside the transmission range of the filter and therefore suppressed.

As the result of the steps performed by the apparatus 15 to 21 inclusive, it will be seen that the consonant frequencies in the range 1250 to 3250 cycles per second have been stepped down in frequency to occupy the range 250 to 2250 cycles per second.

When relay 14 attracts its armature, in response to vowel components, circuit 12 is connected to the input of filter 22 which passes the range 250 to 2250 cycles per second directly to the outgoing circuit 23. After suitable amplification, the currents in circuit 23 are transmitted by way of hybrid coil H_2 into the outgoing line L_2 .

The apparatus in the receiving side R is closely analogous to that in the transmitting branch T and the description will follow readily from that which has been given for branch T. The amplifier detector 25 corresponds to 13 and the marginal relay 26 corresponds to 14. The filters 27, 30 and 33 respectively may be identical respectively with filters 21, 18 and 15. The modulators 28 and 31 may be identical with 16 and 19 as may also the sources 29 and 32 with respect to the sources 20 and 17. In fact, in practice, instead of two sources 17 and 32, one source would probably be used to supply both modulators 16 and 31, and similarly, one source would be used in place of the two sources 20 and 29 shown on the drawings. Filter 34 may be identical with filter 22.

The operation of the circuit of Fig. 3 is as follows. Speech waves coming in over the subscriber's line L_1 pass through the volume control 10 and are reduced under varying conditions of service to the same volume level. These waves pass in part into the branch 12 and in part into the branch 11. If the waves

in the branch 11 after amplification and detection are of insufficient amplitude to cause marginal relay 14 to attract its armature, the branch 12 remains connected to the filter 15.

5 As explained above, the relay 14 is adjusted so that it attracts its armature in response to vowel and semi-vowel sounds but does not attract its armature in response to consonant waves in the circuit 11. Consequently the
10 consonant components in the subscriber's speech occupying the range from 1250 cycles to 3250 cycles per second pass freely through the filter 15 and into the modulator 16 where they modulate the sustained waves of 20000
15 cycles from the source 17. The upper side band resulting from this modulation is passed through the filter 18 to the demodulator 19 where it is combined with the sustained wave of 21000 cycle frequency from
20 the source 20. The lower side band of this demodulating process lies in the range from 250 to 2500 cycles per second and therefore passes freely through the filter 21 and through the outgoing amplifier into the hybrid coil H_2 and the outgoing line L_2 . As
25 the result of this process the consonant sounds which normally occupy the range from 1250 to 3250 cycles per second pass into the outgoing line L_2 in the reduced frequency range extending from 250 to 2250 cycles per
30 second.

Vowel sound components and semi-vowel components present in the subscriber's speech passing into the circuit 11 cause actuation of relay 14 and shift the circuit 12 from
35 the input of filter 15 to the input of filter 22 so that these vowel sounds and semi-vowel sounds which occupy the range from 250 to 2250 cycles per second pass directly into the outgoing branch 23 and so to line L_2 .
40

Since normally each word of speech will comprise partly vowel or semi-vowel sounds and consonant sounds, the relay 14 will be operating continually during sustained
45 speech and for this reason its operation should be made as fast as practicable in order not to seriously clip any portions of the words or syllables being spoken.

From the foregoing description it will be evident that the vowel and the consonant sounds transmitted to the line L_2 both occupy the same frequency range and it is necessary therefore to separate the vowel sounds from the consonant sounds in order to receive the
50 speech in intelligible manner.

The station at the opposite end of the line L_2 may be an exact duplicate of the station shown on the drawing, so that the manner in which the speech waves are received at the
55 distant terminal will be evident from considering that speech waves are coming in on the line L_2 to the station shown in the drawing after having been treated at the distant station in exactly the same manner as has
60 been described above.

It is assumed in the circuit of Fig. 3 that whatever amplification has been made in the system, the consonant sounds received over the line L_2 bear a definite amplitude relation to the vowel sounds, which for convenience
70 will be assumed to be the same as in normal speech. Consonant sounds, therefore, arriving over the line L_2 and passing into the amplifier detector 25 are of too low intensity to cause relay 26 to attract its armature. These
75 sounds therefore, pass directly into the filter 27 and from there into the modulator 28 where they are stepped up in frequency by combining with the wave of 21000 cycle frequency from source 29. The upper side band
80 resulting from this modulation is passed through the filter 30 and is demodulated with waves from the source 32 of 20000 cycle frequency producing a side band extending from 1250 to 3250 cycles per second. This side
85 band is transmitted to the filter 33 and into the line L_1 . As the result of the modulating action just described, it is seen that the consonant sounds which were transmitted at reduced frequency level over the line L_2 are
90 elevated to their normal frequency level before they are transmitted into the subscriber's line L_1 .

Vowel and semi-vowel sound components received over the line L_2 actuate the relay 26
95 and connect the incoming circuit to the filter 34 instead of to the filter 27. These vowel and semi-vowel sound components occupying the range 250 to 2250 cycles per second pass directly into the outgoing subscriber's
100 line L_1 .

In the circuit of Fig. 3 it will be seen that a saving in the total frequency range of 1000 cycles per second has been effected since the components in the range 250 to 3250 cycles
105 per second have been transmitted in the total range of 250 to 2250 cycles per second over the line L_2 .

In the circuit of Fig. 3 the switching at the receiving station was made in the same manner as at the receiving station, that is, depending on the strength of the received sound components.
110

In some instances it may be desirable to synchronize the switching at the transmitting and receiving stations by means of a pilot channel. Such an arrangement is shown in Fig. 4 which will now be described.
115

In the circuit of Fig. 4, the lines L_1 and L_2 with their hybrid coils and balancing network are shown exactly as in Fig. 3. The transmitting branch T is shown as comprising frequency selecting and shifting circuits indicated generally at 40 which may be identical with those shown in the circuit of Fig. 3. Also, in the receiving branch, the frequency selecting and shifting circuits generally indicated at 50 may be identical with those in the branch R of Fig. 3.
120
125

The switching at the transmitter is carried
130

out under control of relay 14 in the same manner as in Fig. 3. However, the switching at the receiver is effected by a differentially wound relay 52 which is controlled over the pilot wire 51. The pilot circuit may comprise the wire 51 and any suitable return, either a ground or another conductor or it may comprise a suitable channel utilizing a separate frequency range of L_2 , by connecting 51 to the upper side of L_2 through a suitable filter. In the present instance, the return for the pilot conductor 51 is the lowermost conductor of the line L_2 .

In the operation of the circuit shown in Fig. 4, speech received over the subscriber's line L_1 , after passing through the volume control 10, passes into the frequency selecting and shifting circuits generally indicated at 40. If the sounds at any instant under consideration are consonant sounds, relay 14 does not attract its armature but these components pass into the circuit 40 where they undergo a frequency shift and then pass out onto the line L_2 in the manner described in connection with Fig. 3.

In the case of vowel and semi-vowel sounds, however, relay 14 attracts its armature so that the waves representing the vowel components pass directly to circuit 40 and into the outgoing line L_2 without change in frequency level. Relay 14 is, however, in the circuit of Fig. 4 provided with a second armature 53. When relay 14 is actuated in response to vowel sounds, armature 53 closes a circuit which extends from battery 54 through the armature 53 and contact of the relay 14, and thence through a divided circuit. One branch of the divided circuit comprises conductor 55 and the lower side of the line L_2 to the distant station. At that station the lower conductor of the line L_2 is connected to the pilot conductor 51 through a circuit the nature of which will be clear from considering the circuit of Fig. 4. Assuming for the moment, however, that the path has been traced at the distant station to the pilot wire 51, it continues through this wire to the station of Fig. 4, thence through the lower winding of relay 52 and back to the battery 54. The other branch circuit that was mentioned extends from the armature 53 through the artificial line 56 and the upper winding of relay 52 back to the battery 54. The current thus traced through the two windings of relay 52 neutralize each other in their effects upon the armature of this relay so that the armature remains unattracted.

The effect of this pilot circuit at the distant station may be seen by considering that a relay similar to 14 has been actuated at the distant station. The circuit is closed therefore from a battery corresponding to battery 54 at that station through armature corresponding to 53 of the relay assumed to be present at that station, through a conductor

corresponding to 55 at that station, through the lowermost conductor of line L_2 of the station of Fig. 4, thence through conductor 55, artificial line 56, both windings of relay 52 in series and pilot conductor 51 back to the distant station where the pilot conductor leads through the lowermost winding of relay corresponding to 52 at that station and to battery 54, already mentioned, at that station.

From the circuits that have been traced to the pilot conductor, it is obvious that relay 52 does not respond to attractions of armature 53 of relay 14 at the same station but does respond whenever armature 53 of relay 14 at the opposite station is attracted.

As in the case of Fig. 3, when the switching relay 52 in the branch R attracts its armature, the circuit is prepared for the proper reception of the vowel and semi-vowel components. In the case of Fig. 4, relay 52 attracting its armature as described, shifts the circuit to the proper branch in the frequency selecting and shifting circuit 50 so that the vowel sound components are transmitted through to the subscriber's line without frequency change. Consonant components leave relay 14 at the transmitting end unactuated, which consequently cause relay 52 at the receiving end to be unactuated. These components therefore undergo frequency shifts in circuits 40 and 50 at opposite stations, as in the system of Fig. 3.

One advantage in using a pilot conductor is that separate amplifiers 60, 61, 62, 63 may be utilized and may be separately and independently adjusted in the branches of the transmitting and receiving circuits, respectively.

In the case of a very long transmission line L_2 , the high frequency low intensity components are attenuated more severely than the low frequency components and therefore require greater amplification. Also, it may be desirable to amplify before transmission the lower intensity components to enable them to override noise present on the circuit L_2 . Since the actuation of the receiver R is not dependent upon the relative levels of the consonant and vowel sounds as received over the line, the relative amplification of the two paths at the transmitter and at the receiver may be made to suit any requirements as to attenuation, noise, etc.

The amplifying elements 62 and 63 indicated at the receiver are shown as variable. These instead of actually amplifying, may in fact introduce losses in one or the other of the two circuits in order to restore the components to their proper or desired relation in the received speech.

In the example that has been given, an overlap of 1000 cycles in frequency has been allowed for the vowel and semi-vowels on the one hand and consonant sounds on the other,

and the frequency level of the consonant components has been stepped downward to increase the overlap with respect to the vowel sounds. It is within the invention, of course, to shift the frequency level of the vowel sound components to increase the normal overlap with respect to the consonant components.

The invention is not to be construed as limited to particular frequency values or ranges that have been given by way of example, nor to the specific circuits that have been illustrated. For example, if the line L_2 were capable of transmitting the range 250 to 2750 cycles per second, the division might be as follows:

Vowel and semi-vowel sounds—250 to 2750 cycles per second,

Consonant sounds—1250 to 3750 cycles per second.

Various other frequency limits and frequency divisions will occur to anyone desiring to practice the invention in connection with a particular system or situation. The scope of the invention is therefore to be determined from the claims.

What is claimed is:

1. The method of reducing the frequency band width required for a communication channel which comprises separating large energy—low frequency wave components from small energy—high frequency wave components occurring in separate instants of time, and shifting the frequency level of at least one of said classes of components so that they overlap in frequency another of said classes of components when traversing said channel.

2. The method of reducing the frequency band width required for a communication channel which comprises dividing the waves on an energy-frequency-time basis into classes of components occurring in non-overlapping time instants, the components of one class differing both in energy level and frequency range from those of another class, and changing the frequency level of at least one class of components so that they overlap that of another class when traversing said channel.

3. The method of transmitting complex signal waves whose statistical characteristics permit division from instant to instant into amplitude components which also differ in frequency range, comprising dividing the waves on an amplitude basis before transmitting, shifting the frequency level of certain of the subdivided portions to overlap that of others, transmitting in their successive instants of occurrence the components of shifted frequency and the others, again dividing the waves as received at the receiving station, and restoring to their former frequency level those components which were shifted in frequency.

4. The method of speech transmission comprising separating at each instant of time

the vowel sounds from the weak consonant sounds, and shifting the frequency of one such group to overlap that of the other whereby both vowels and consonants are transmitted in a band of frequencies less wide than that occupied by the sounds as produced.

5. The method of speech transmission comprising separating the relatively strong from the relatively weaker components occurring in successive instants of time and shifting the frequency level of one such group of the separated components so that they occupy the level occupied by the other group, whereby both groups of components are transmitted in a band of frequencies less wide than that of the speech components as produced.

6. In a telephone system, means for separating in successive instants of time vowel sound components from weak consonant sound components, means for shifting the frequency of one such group of components to increase their frequency overlap with respect to the other group, whereby both vowel and consonant sounds in successive time instants occupy a narrower frequency band than in normal speech, and means to transmit the resulting wave components to a distance.

7. In a telephone system, means at a sending station acting in response to amplitude changes for separating wave components in successive time instants into separate circuits, modulator means associated with one of said circuits for shifting the frequency level of the components in said circuit to cause them to occupy the frequency range of the components in the other circuit, means for transmitting to a distance the shifted frequency components and the other components, and means at a receiving point for restoring the shifted frequency components to their original frequency level.

8. A system according to claim 7 in which the means at the receiving point for restoring the components to their original frequency level comprises a means for separating into separate circuit branches received wave components representing large amplitude components in the original speech from those representing relatively weak components in the original speech waves, and modulator means associated with one circuit branch for shifting the frequency level of the components therein to their original frequency level.

9. In a telephone system including transmitting and receiving stations and a transmission path interconnecting them, a pair of circuit branches connected to said path at a transmitting station, means for switching into the first circuit branch the low-frequency large-amplitude components of impressed speech waves, and for switching into the second branch the high-frequency low-amplitude components, modulating means in at least one of said circuit branches for shift-

ing the frequency level of the components in said branch to occupy the level of the components in the other branch, similar circuit branches at a receiving station, including switching means for separating respective components received over said path into said circuit branches, and a pilot channel inter-connecting said stations for synchronizing the operation of the switching means at said respective stations.

10. A system according to claim 9 including at the transmitting station means to increase the amplitude of the low amplitude components in said second circuit branch with respect to the components in the first branch, prior to transmission of the components over said path.

11. In a telephone system, means for conserving the total frequency band width used for transmission, comprising marginally operating switching mechanism and frequency shifting mechanism, for first separating vowel components from weak consonant components occurring in successive times and shifting the frequency level of the components of one of said bands so that they have a greater frequency overlap with respect to other components than they do in normal speech.

12. A system according to claim 11 including amplifying means for separately amplifying the consonant components to any desired extent with respect to the vowel components before transmission, to compensate for the higher attenuation of the consonant components by said system.

13. A system according to claim 11 including means to provide an increased speech signal-to-noise ratio comprising means for transmitting the weaker sound components with a greater increase in amplitude level than the stronger components.

In witness whereof, I hereunto subscribe my name this 16th day of October, 1930.

JOHN C. STEINBERG.