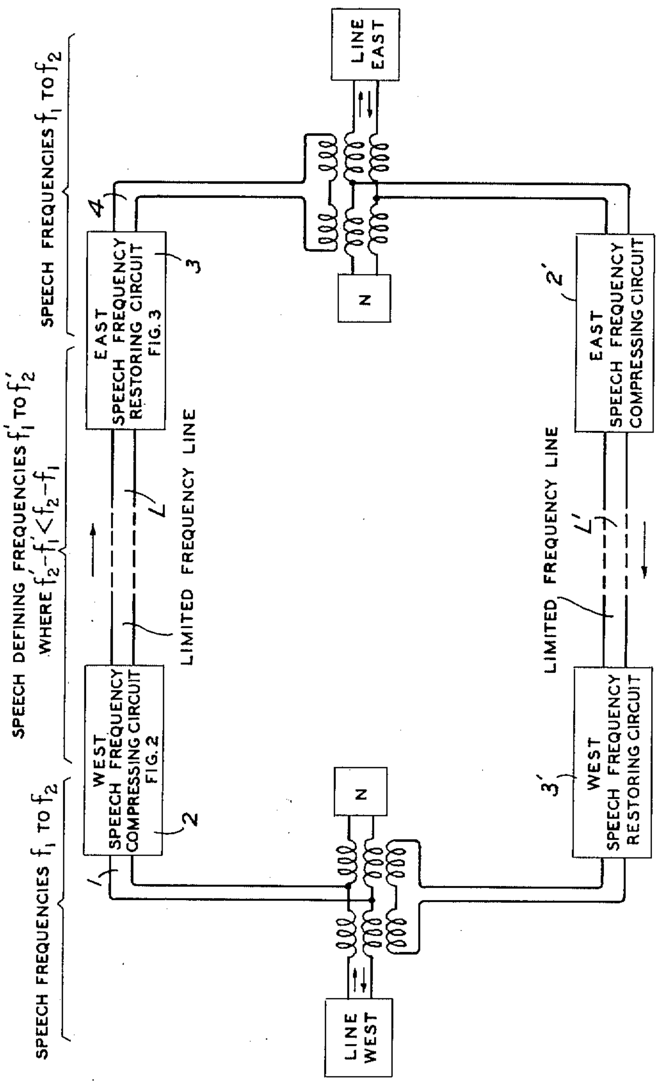
Nov. 16, 1937. H. W. DUDLEY 2,098,956

SIGNALING SYSTEM

Filed Dec. 2, 1936 7 Sheets-Sheet 1

FIG. 1



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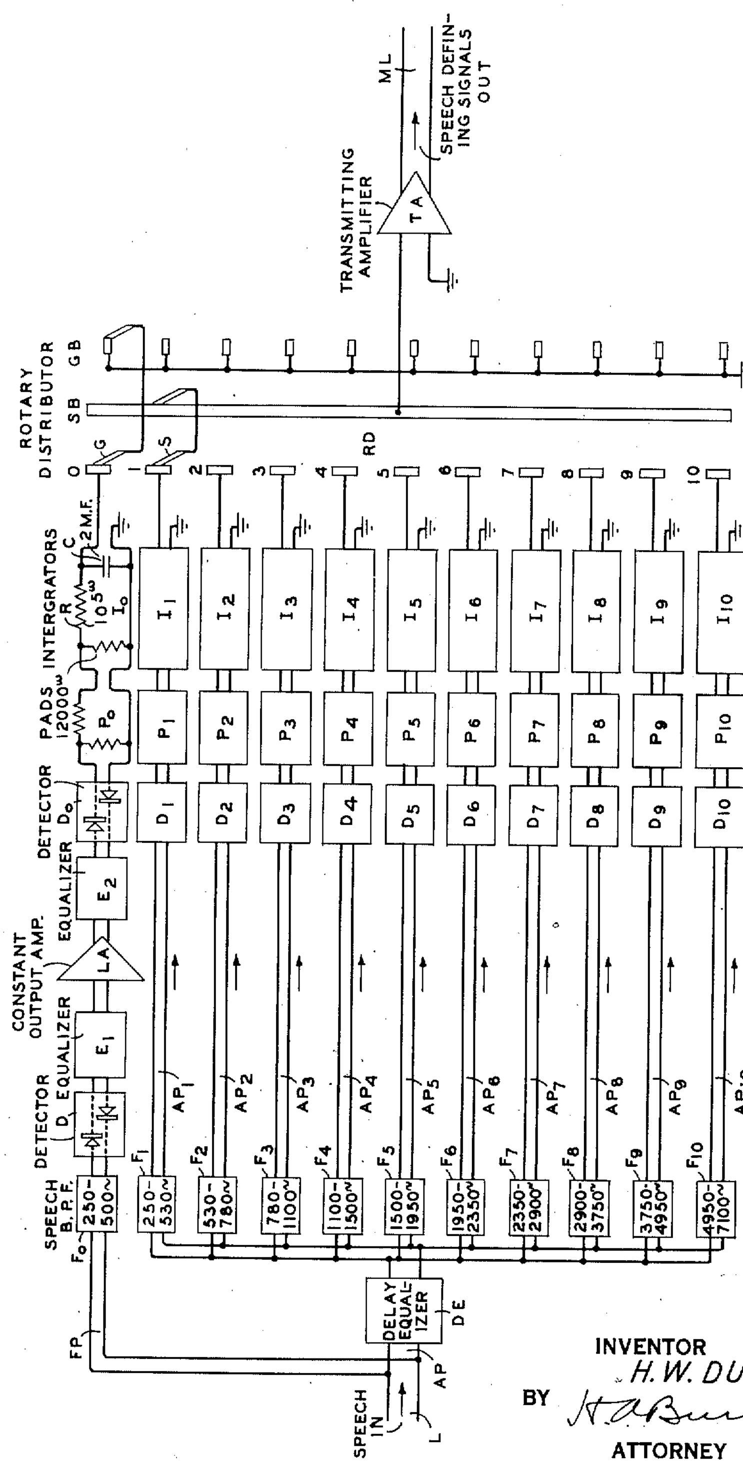
SIGNALING SYSTEM

Filed Dec. 2, 1936 7 Sheets-Sheet 2

FIG. 2



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Filed Dec. 2, 1936 7 Sheets-Sheet 3

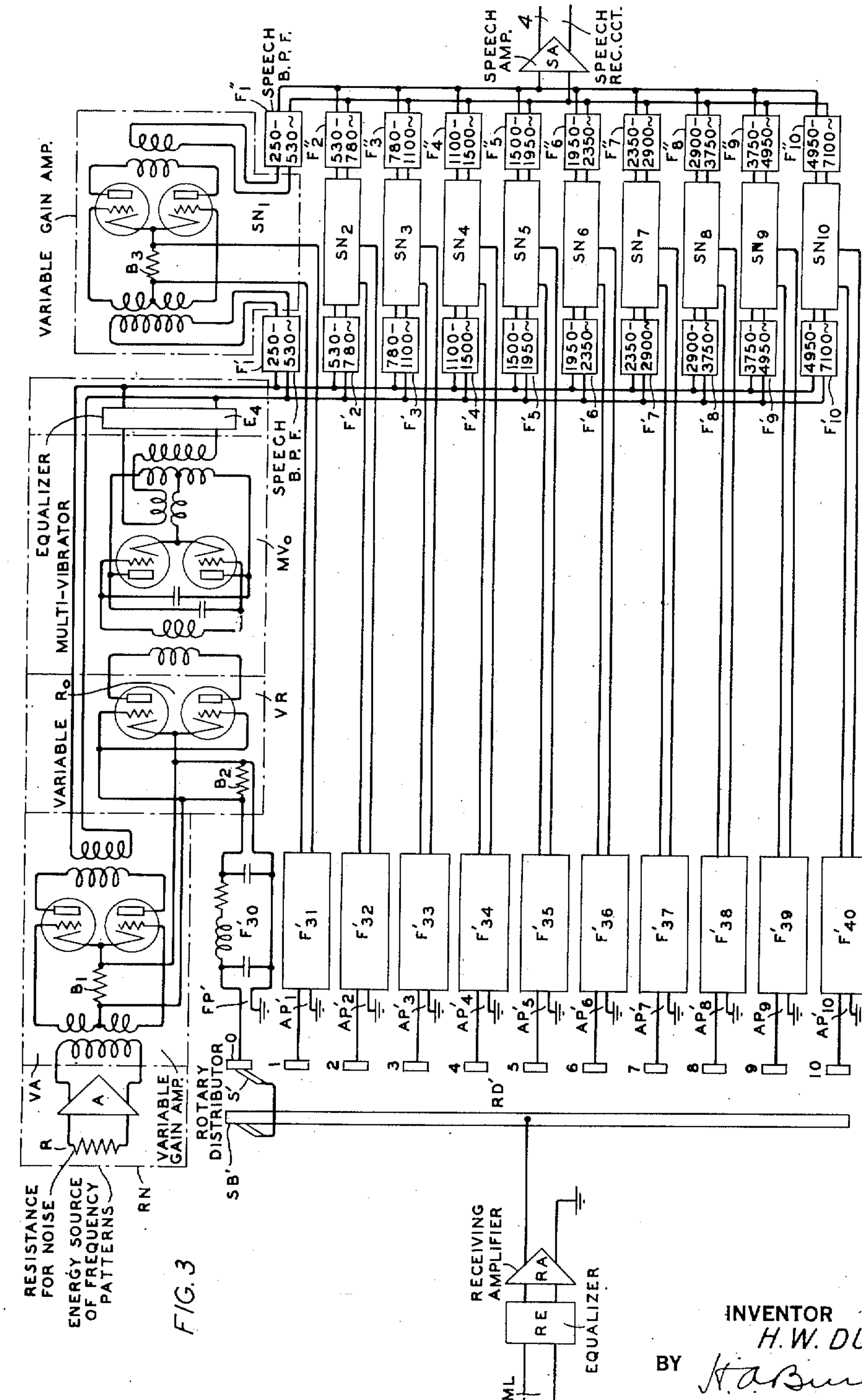

FIG. 3

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SIGNALING SYSTEM

Filed Dec. 2, 1936 7 Sheets-Sheet 4

FIG. 4

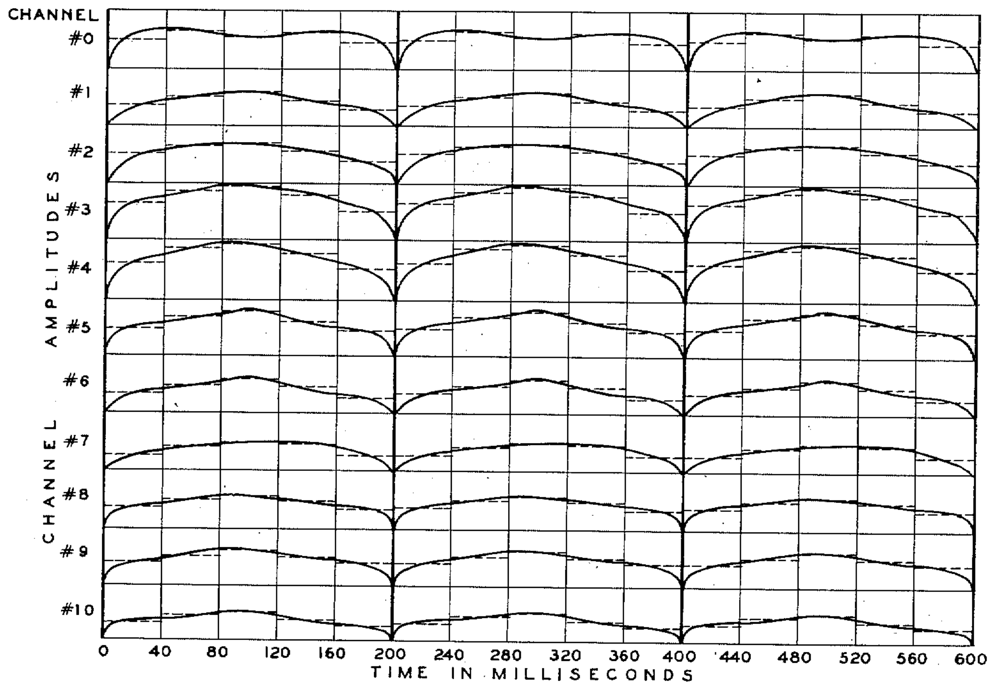


FIG. 5

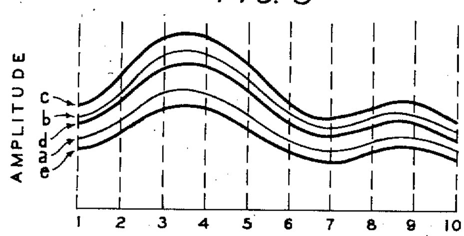


FIG. 6

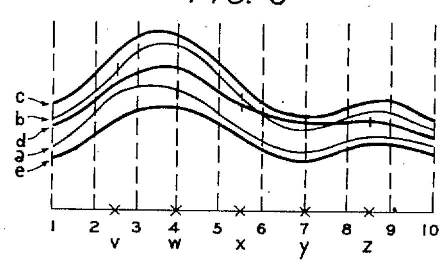
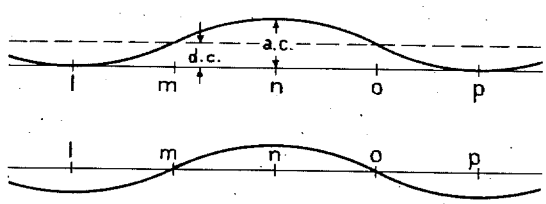


FIG. 7



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SIGNALING SYSTEM

Filed Dec. 2, 1936 7 Sheets-Sheet 5

FIG. 8

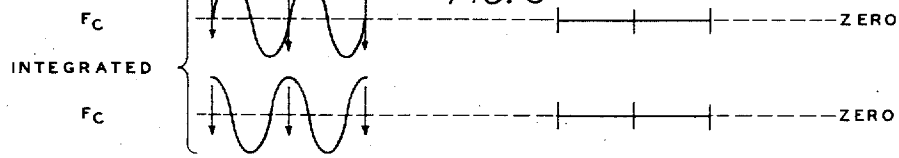


FIG. 9

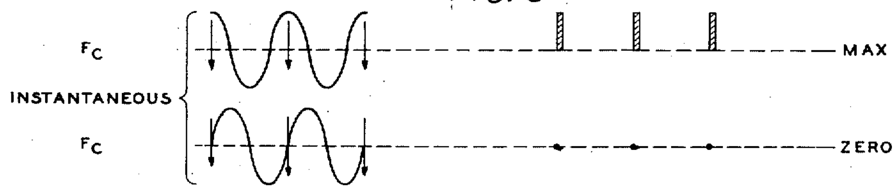


FIG. 10

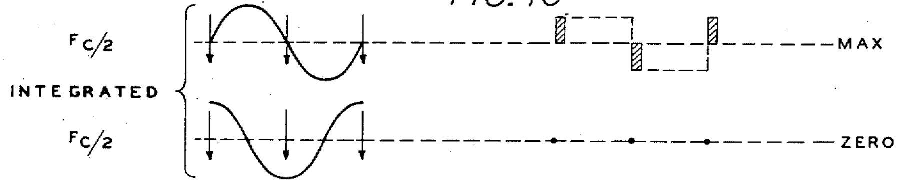


FIG. 11

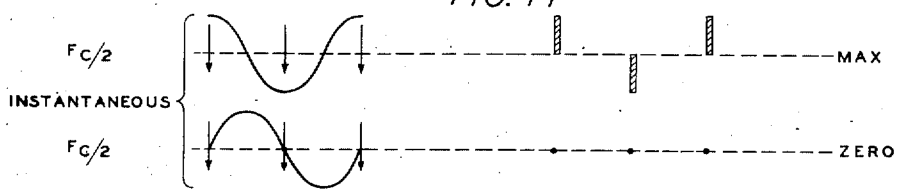


FIG. 12

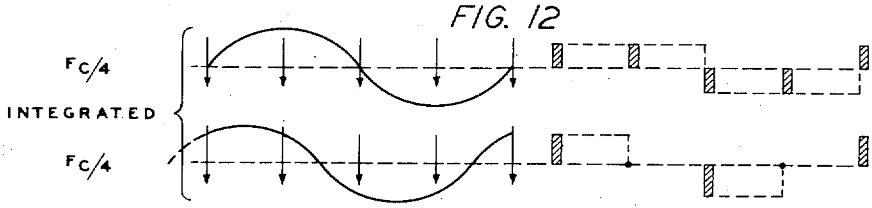
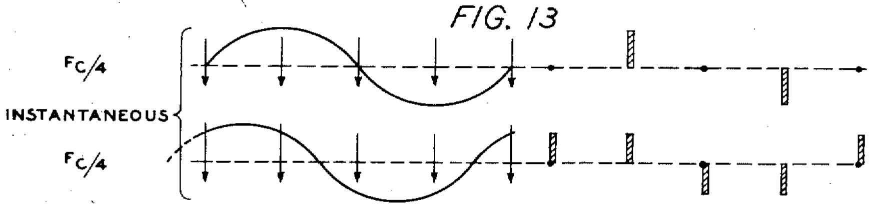


FIG. 13



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SIGNALING SYSTEM

Filed Dec. 2, 1936 7 Sheets-Sheet 6

FIG. 14

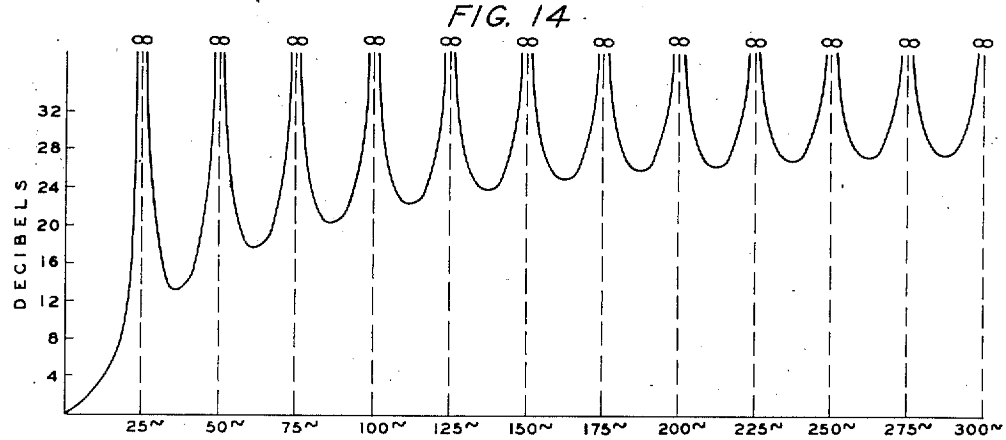


FIG. 15

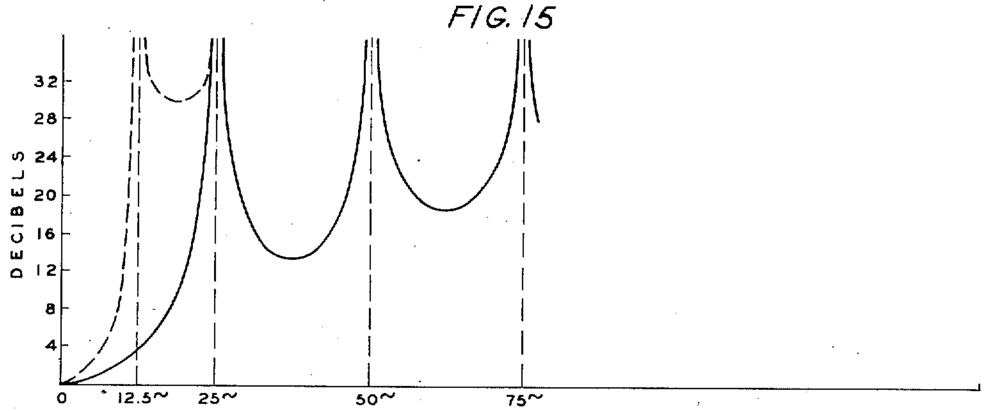
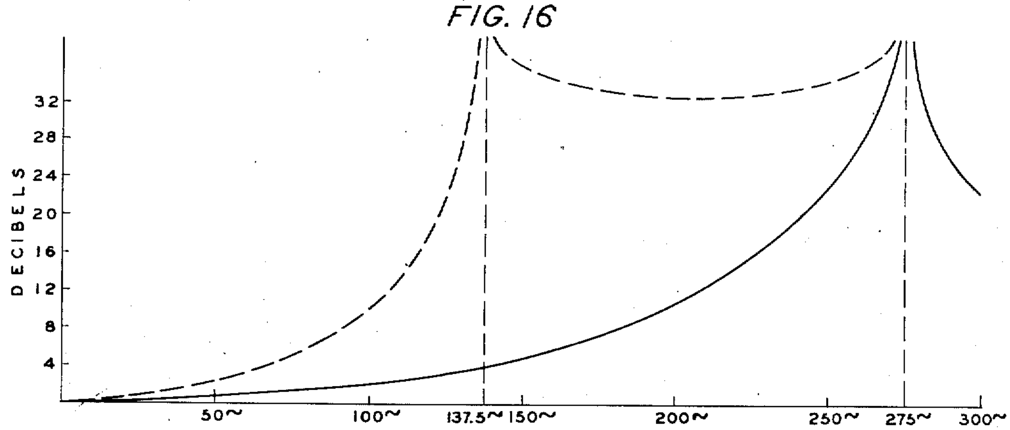


FIG. 16



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2,098,956

SIGNALING SYSTEM

Filed Dec. 2, 1936

7 Sheets-Sheet 7

FIG. 17

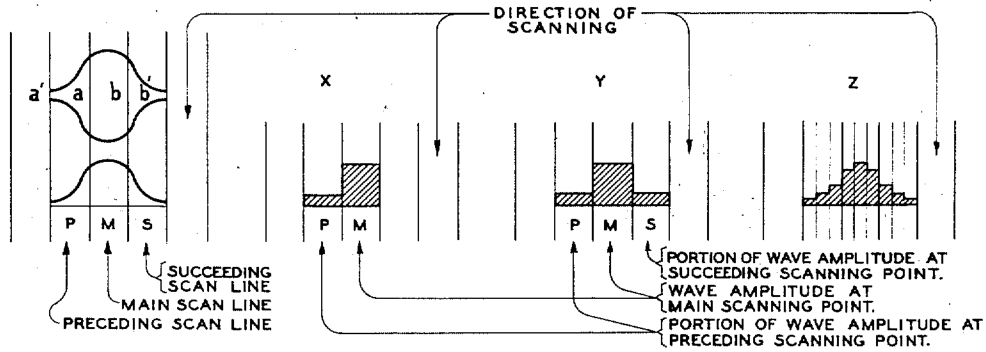


FIG. 18

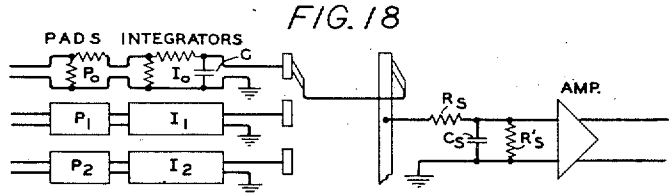


FIG. 19

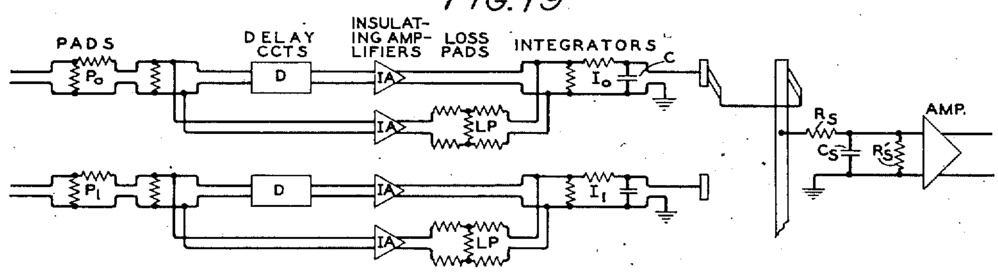
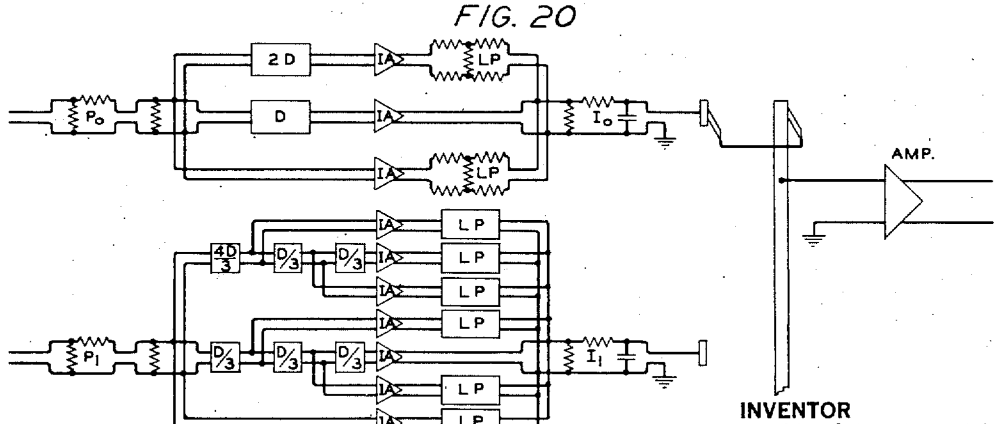


FIG. 20



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UNITED STATES PATENT OFFICE

2,098,956

SIGNALING SYSTEM

Homer W. Dudley, Garden City, N. Y., assignor to Bell Telephone Laboratories, Incorporated, New York, N. Y., a corporation of New York

Application December 2, 1936, Serial No. 113,818

27 Claims. (Cl. 179—1.5)

This invention relates to wave transmission and signaling systems.

An object of the invention is to reduce the frequency range required for the transmission of signals such, for example, as speech signals.

A further object of the invention is to effect such reduction without necessitating increase in the time of transmission.

It is also an object of the invention to facilitate transmission of a message over a transmission medium not adapted to readily pass the frequency band originally occupied by the message as, for example, to facilitate telephoning over long submarine cables where the higher speech frequencies cannot readily be transmitted because of excessive attenuation.

Another object is to facilitate increasing the number of successive channels that a wave transmission medium such as, for example, a land line or radio circuit, can transmit in a given frequency range.

Still another object is to increase the signal-to-noise ratio obtainable at the receiving end of a system without increasing the sending level, as for instance, in long distance radio communication where the transmitting power required tends to become excessive.

It is also an object of the invention to promote secrecy in transmission of signals, for instance, in wire or radio transmission of speech.

The information transmitted by speech does not absolutely require all the frequency space allotted to it in the human voice. A specific case can be worked out as to how much frequency band is required as a minimum, for example, by determining and taking account of the number of the independent variables or parameters involved in speech production (i. e. the number of the independently movable physical elements of the vocal system that are involved in speech production) and the rate at which these vary. This is done hereinafter, and from the computation it is estimated that speech from 250 to 7100 cycles can be represented almost perfectly by the information that can be passed in a band having a width of the order of 500 cycles, using the same time of transmission as the speech itself requires.

Analyzing the vocal system of a man from the broad viewpoint of producing speech sounds, it is seen to be made up of two types of parts, (1) fixed and (2) variable. The variable parts are considered here to be those that vary in position from sound to sound. Examples are the lips and teeth opening and closing, the tongue shifting forward and backward, the vocal cords varying in tension, and the uvula opening and closing the nasal passage. The term "fixed" is here used in its broadest sense. It not only includes parts that are not moved from sound to sound in speech, such as the nasal passages, pharynx and much of the larynx, but it also includes any fixity of feature. As an example, the fact that the vocal cords are always used in the voiced sounds is a fixed feature as is also the fact that they always vibrate in the same buzzer-like way as regards the presence of a fundamental frequency and all of its overtones up to a large number greater than 30; the variation of the fundamental frequency, or pitch, of the vocal cords is, on the other hand, a variable feature as stated previously.

The whole vocal system may be likened to a mechanical-acoustical oscillator with certain fixed circuits and certain variable mechanical elements. The part of the vocal system corresponding to the fixed circuits of the oscillator is the same from man to man. It is the same from sound to sound in the same man with the different elements taking on different values to produce the different sounds. To make the analogy just referred to more specific, consider that the vocal system is, in principle, like the ordinary electrical oscillator mounted in a box as a fixed piece of apparatus, the variability being obtained by switches for starting the oscillator and for choosing the desired inductances, by continuously variable dials for selecting the capacitance, and by step variable dials for adjusting the resistances controlling the output. With such an arrangement other features such as feedback may also be controlled.

Just as the oscillator when oscillating is still essentially a fixed piece of apparatus to which variable controls of frequency, output and feedback, are applied, just so the fixed vocal system includes the condition of the vocal cords vibrating at an average or other specified steady rate, to which condition the controls for varying or modulating the generated vibrating signal in the production of speech can be applied. The importance of including a normal vibration of the vocal cords as a fixed feature is due to the fact that they vibrate on the average at a fundamental frequency of 100 to 150 cycles per second, whereas the variable controls specified can change only at rates ten or more times smaller than this. Strictly, the fixed features include what may be described as a multioscillator source of energy rather than a single one, for not only are there the periodic oscillations produced at the vocal cords, but there are also non-periodic or random oscillations produced by the passage of air through restricted openings such as between the lips and teeth for the "f" sound, between the tongue and hard palate for the "sh" sound, between the vocal cords themselves for whispering, etc.

This differentiation between fixed and variable features also characterizes the type of modulated speech signal produced. In this case the fixed features correspond to the sustained oscillatory sound produced with the various elements or parts of the vocal system in an average or normal position. This means an average lip opening, an average vocal cord tension, etc. The variable features correspond to the changing or modulating of the sound by varying the different elements from their average positions. It will be clear, therefore, that the fixed features appearing in a speech signal are oscillatory in nature and the variable features are modulatory.

An analysis of an oscillogram of a speech wave shows that there are variations from maximum swing in one direction to maximum swing in the opposite direction in .001 second or less. Yet these are oscillatory swings for in the next period about .010 second later the same swing will still be found as an almost identical copy. If the slight change that occurs from period to period is followed up until it becomes great enough so that the original wave form is lost, it will be found to occur many periods later, oftentimes twenty or more, requiring a time of the order of .200 second. This latter type of change is the modulatory type of change. It is easiest seen in a single sound as the building-up of the peak amplitudes to a maximum and then a falling-off to zero again. In this case a complete change corresponds to a large part or all of a whole syllable and such change is therefore known as a syllable or syllabic frequency change.

From the foregoing it is evident that speech has a dual characteristic. On the one hand we have fixed parts or elements setting up oscillatory waves containing relatively high frequency patterns. On the other hand we have varying parts or elements setting up modulatory waves of low syllabic frequency pattern. An ideal frequency range compressing system would be one that took advantage of the dual nature of speech by setting up at the receiving end all of the fixed features and transmitting over the line from the sending end only information as to the instantaneous positions of the variable parts. In such case it would not be necessary to transmit information as to each fixed feature.

The frequency range compressing system need not be so identical a copy of the vocal system however. It is well known that one set of parameters can be substituted for another without any loss of definition so long as the number of independent parameters remain unchanged. Any change from the simple ideal mentioned does generally lead to a larger number of required parameters because the newly selected ones are not independent. However, this is not of much practical importance, for the great saving of frequency range accomplished by the invention is obtained by transmitting only syllabic frequencies corresponding to the variable or modulatory controls of the speech signal, without transmitting the high frequency pattern corresponding to the fixed modulatory characteristics. As this saving is of the order of 10 to 1, an increase of 50 per cent to 100 per cent in the parameters required to satisfactorily represent speech, is unimportant.

As pointed out in detail hereinafter, the number of independent variables involved in the production of speech is small. That is, the number of movable or variable elements of the vocal system that are controlled as parameters to give the desired speech production, and are movable or variable substantially independently of one another by the muscles of the vocal system, is small. In other words, the number of variables or parameters that can be controlled substantially independently in speech production is small, being of the order of ten. Moreover, as indicated above and discussed hereinafter, for each of the physical elements the minimum time in which it can go through a complete cycle of change in position is not less than one-tenth of a second. Consequently, each independent variable has a fundamental frequency of not over ten cycles per second, while engaged in speech production. Further, as explained hereinafter, a frequency band having a width not more than two or three times greater than the product of this frequency and the number of the independently variable physical elements can suffice to transmit signals containing practically precise information as to the instantaneous displacement values or positions of the independently variable physiological elements.

Such a band, therefore, contains the variant or unpredictable information in the speech, and so constitutes speech defining signals that define the variable characteristics of the speech. As explained before, these defining signals may be any signals derived from speech signals provided the derived signals give as many independent variable quantities or parameters as the number of independent variables involved in the production of speech. Furthermore, the chosen parameters need not be entirely independent provided their number be increased sufficiently to make up for their lack of independence. For example, if the original speech band be divided into a sufficient number of sub-bands the chosen parameters may be merely the average amounts of power in the several sub-bands, as brought out in detail hereinafter.

There exists, then, in the actual production of a complex wave by the vocal system, a simple set of slowly varying elements or parameters (the independent variable elements referred to above of the vocal system), that determine the variable characteristics of the signal which are referred to above. And, to transmit information that will suffice for defining or reproducing the variable characteristics it is unnecessary to transmit the fixed characteristics of speech, it being sufficient to transmit information defining the variations of any simple set of parameters derived from the complex speech wave and corresponding to the independently variable elements of the vocal system as regards number and independence, or as regards the number of degrees of freedom of variation.

Like considerations hold in the case of other complex signals, such as music, which, as in the case of speech, have a sluggish rate of change modulating or controlling a rapid rate of change of signal strength or, from the point of view of frequency, a set of low frequencies modulating a set of high frequencies.

In accordance with one feature of the invention, to transmit a complex signal, such as speech, variant information regarding the variable or unpredictable characteristics of the signal is transmitted, instead of the complex signal wave. The waves transmitted can define the signal precisely, as regards its unknown or variable characteristics, yet have small frequency range relative to the signal wave, and at the same time be as short in duration as the signal. The signal defining waves transmitted need not directly contain any intelligibility whatsoever. Thus, a high degree of secrecy is attainable.

In accordance with a feature of the invention as applied to the transmission of speech, for example, advantage is taken of the fact that much of the information ordinarily transmitted is of an invariable or predictable character due to the general uniformity of the speech producing organs from person to person, by reproducing such predictable information artificially at the receiving end of the transmission system in order that it need not be transmitted from the sending end. Thus, effective use is made of information or foreknowledge of the fixed or invariable characteristics of the signal source, with the result that the frequency band width of transmission can be reduced.

In one specific aspect, the invention is a system in which a speech signal is analyzed for its fundamental frequency, and for the average power in properly chosen sub-bands of frequency, this information being transmitted and then used at the receiving end to fashion waves from a local multifrequency source into a simulation of the signal.

To fashion the simulation of the signal from the waves supplied from the local source, frequency sub-bands of these locally derived waves are selected which are, respectively, coextensive with the chosen sub-bands of the speech signal, and the average power in each sub-band of the locally supplied waves is varied in accordance with the power in the corresponding chosen sub-band of the signal wave. This variation is effected in response to the information transmitted from the sending end of the system regarding the average power in the chosen sub-bands of the signal wave.

Two types of frequency spectrum are used alternately in speech, (1) a continuous spectrum in the case of hissing or unvoiced sound, and (2) in the case of voiced sounds a discrete spectrum with a variable fundamental and with upper harmonics to a relatively high frequency always present. Hence, the local source provided, preferably is such that the waves supplied by it can have either type of spectrum. The type is determined in response to the information transmitted from the sending end of the system with regard to the presence or absence of a fundamental frequency in the speech wave and the magnitude of any such fundamental frequency. In other words, if the fundamental frequency is present the discrete spectrum is generated by the local source, and if no fundamental frequency is present a continuous spectrum is generated.

Significant changes in the fundamental frequency of the speech sounds and in frequency distribution of power in speech can take place only at a rate which is limited by the sluggishness of the muscles of the vocal system to less than about ten cycles per second, (a frequency much lower than the fundamental oscillatory frequencies of vocal cords which range from about sixty cycles to in the neighborhood of five hundred cycles). It therefore results that the equipment required at the sending end of the system for analyzing the speech signal as to its fundamental frequency, and likewise the equipment provided at the receiving end of the system for responding to the transmitted indications as to the fundamental frequency of the speech sounds, need only be responsive on the line side to frequencies up to perhaps one to three times the frequency of ten cycles per second, just mentioned, depending on the accuracy desired in the transmission of the indications.

Moreover, the number of sub-bands analyzed for power content need not exceed five or ten, for example, to obtain high intelligibility; because as indicated above and pointed out in detail hereinafter, the number of independent variables or parameters in speech is small, and the power in each sub-band is largely independent of that in the others, particularly as the distance between the mid-frequency bands is increased.

Such a system, then, analyzes the signal as to its fixed features and variable features. The fixed features include; (a) the existence of definite frequency sub-bands in which the power distribution is sensibly uniform, (b) the existence of a frequency spectrum that alternates from the continuous type of spectrum to a discrete type with varying fundamental and with all upper harmonics always present, and (c) the fact that time variations of the fundamental frequency and of the power in the frequency sub-bands occur only at syllabic frequency rates. The variable features include; (A) the magnitude of the average power in each sub-band, and (B) the nature of the signal spectrum (as to whether it is continuous or discrete and, in the latter case, as to what is the magnitude of the fundamental frequency).

Since there is foreknowledge at the receiving end as to the fixed features or characteristics of the signal, they can be supplied locally at the receiving end and it is unnecessary to transmit information regarding them. Their supply locally is accomplished by the choice of the type of circuit, the choice of elements to simulate the vocal cords and the eddying constrictions of the vocal system, and the choice of frequency sub-bands. It is now sufficient to transmit information defining the variable characteristics and combine them with the locally supplied fixed circuit features to reproduce the signal.

In my application, Serial No. 47,393, filed Oct. 30, 1935, these various kinds of speech defining signals are transmitted by modulating them upon low frequency carriers and then detecting from the carriers at the receiving end the modulating currents which are then used to control the local sources of oscillations to reproduce speech. In accordance with the present invention, however, I propose to transmit the speech defining signals corresponding to the several different channels of desired information, by using rotary distributors or commutators which successively transmit to the line the instantaneous values of the waves in the several channels one after the other, at such a speed of commutation that a sufficient number of the values of the wave in each channel will be transmitted to fairly represent the wave. At the receiving end a similar distributor or commutator distributes from the line to the corresponding information channels, the wave elements transmitted over the line. It will be understood, of course, that the arrangements herein shown are illustrative only, and that equivalent scanning, commutating, or distributing devices may be employed.

In a system of the type shown in my application, Serial No. 47,393, above referred to, in which the speech defining signals are transmitted by carrier waves, a considerable number of filters are necessarily used for selection between channels. Consequently, the part played by phase, in the terms of delay and delay distortion, becomes important. By delay, of course, is meant the lag introduced by the various circuit elements in the time of arrival of the wave at the given receiving point as compared with the time it starts from the sending end. This may be thought of as a phase change. The delay distortion referred to arises from the fact that the different frequency components are delayed by different amounts and, hence, arrive at the receiving point in different relative phase relations from those existing at the time the several frequency components started from the sending end.

Delay is important in long two-way speech circuits, such as the case of the proposed transatlantic telephone cable, where the lines connected to the ends of the cable would have so much inherent delay because of their great length that very little additional delay could be tolerated in the apparatus associated with the ends of the cable. Delay distortion, on the other hand, is very important because if it is not compensated it changes aspects of the signals so as to impair them, and, on the other hand, if correction is made for delay distortion it could be done only at great expense both of money and of increased delay.

A low frequency carrier system of the type disclosed in my application, Serial No. 47,393, requires a set of band-pass filters at the transmitting end and another set at the receiving end to separate out the different channels of speech-defining signals. These filters, of course, introduce delay into the circuit. While they are necessary in the carrier system to maintain separation between the several channels they are unimportant so far as the actual compression of speech signals in the cable or other transmitting medium is concerned.

To illustrate their relative unimportance in this connection, consider the following case. Suppose we have eleven speech signaling channels of say 10 cycles each. Let us transmit these channels over eleven separate lines instead of over the main line. Then each line needs to be good only to 10 cycles to constitute a satisfactory transmission medium. The total frequency space required for the ten lines then is 11×10 cycles or 110 cycles. But these eleven channels on the same line will require more than a band of 110 cycles to allow for frequency selection, and will, in addition, involve an appreciable delay due to the filters employed. If, however, instead of using carrier channels, rotary distributors or commutators are used as now proposed, the delay introduced by the filters can be eliminated by synchronizing the distributors at the sending and receiving ends so that at the receiving point we will know when a given wave element is received that it was sent from the sending end a certain number of microseconds earlier. Using the terminology of picture transmission we can "frame" and then keep in synchronism so that each bit of information received is sent to its proper channel.

The commutator or distributor used at each end of the circuit may include rotary wipers passing over segments, one corresponding to each signal determining channel. The commutator brush will rotate at a speed sufficiently high to give the required information about the various voltage changes representing individual waves. The minimum speed of rotation would be about twenty times per second and, assuming there are eleven channels, this would involve commutation over 220 commutator elements per second. Higher speed will, of course, give better definition. Thus a speed of 50 revolutions per second or 550 commutator elements would give a much improved quality.

This latter speed corresponds to a fundamental frequency to the line of one-half 550 or, in other words 275 cycles per second. This is because, for the worst conditions, we may have a voltage in one direction on the commutator element of a given channel during one commutation; and during the next commutation the voltage may be reversed. Considering then the frequency range of 275 cycles as the range which is required to be transmitted, the line would have to be able to transmit a slightly wider range, say from 10 per cent to 40 per cent wider, to allow for the fact that the commutator system (which, as will be pointed out later, acts as a filter) will not have a sharp cut-off. Therefore the total frequency range on the line might be 1.4 times 275 cycles or 385 cycles. This should give at the receiving end a very good restoration of the speech signal. If the band transmitted over the line should be extended to 800 cycles, the system would give very nearly perfect reproduction.

As indicated above and hereinafter, the invention is by no means limited to the specific form herein disclosed.

Other objects and aspects of the invention will be apparent from the following description and claims:

Fig. 1 shows schematically a system embodying the invention in the specific form referred to above;

Fig. 2 is a detailed showing of the transmitting circuit at the west terminal of the system, this transmitting circuit being the west speech frequency compressing circuit shown as the block in the upper left-hand corner of Fig. 1;

Fig. 3 is a detailed showing of the receiving circuit at the east terminal of the system, this receiving circuit being the east speech frequency restoring circuit shown as the block in the upper right-hand corner of Fig. 1;

Figs. 4 to 17, inclusive, are curves showing various features of the operation of the system; and

Figs. 18, 19 and 20 show alternative arrangements of certain parts of the system.

There is disclosed hereinafter a particular circuit which can be used for reducing the frequency range of speech by setting up artificial currents of limited frequency range which will transmit information about the independent variables of the speech producing organs in man. The first important question is: What frequency range does speech inherently require for transmitting its information content?

In the operation of such a system with high quality transmission of speech, the person at one end of a telephone line talks into a high quality subset. At the distant end the listener also has a high quality subset. The transmission between the two is not direct but is carried on in the following way:

At the sending end is special apparatus to tell the characteristics of the speech signal being fashioned by the talker. At the other end is receiving apparatus to receive these signals and produce speech that is a very close copy so far as the ear can determine of that at the sending end. In between is a transmission line of limited frequency range. The transmission must take place as rapidly as the speech is produced. It is desirable to determine roughly what is the minimum frequency band that can be used on the transmission line for such high quality transmission.

A rough answer to this problem can be obtained by the following method:

We determine how many independent variables are involved in the production of speech. Next we determine the minimum amount of time in which any one of these variables can go through a complete cyclic change of position. From these the frequency band required for the transmission of reasonably precise information as to the instantaneous position of each independent physical variable is obtained. Then, next, multiplying this frequency band by the number of independent variables we have the total minimum frequency band required. For actual transmission in circuits allowance must be made for the inefficiency of filter or other auxiliary circuits that will be required.

It is convenient to differentiate the production of vowel sounds from those of consonants. As used here, vowels are taken to indicate the pure vowels, the semi-vowels, the diphthongs, and the transitionals. Some thirty-four of these are listed in the book "Speech Pathology" by Lee Edward Travers. They comprise fourteen vowels as, for example, a in art; eleven diphthongs as, for example, a in mat; five semi-vowels as, for example, m in me, and four transitionals as, for example, wh in white. These vowel and near vowel sounds are set forth in detail in my application, Serial No. 47,393.

These vowel and near vowel sounds have, in general, been produced artificially so that we have a very good check on the number of parameters involved in producing them. From the vocal standpoint, starting at the lungs and coming out to the mouth opening, the variables that may be controlled independently are:

1. Lung pressure
2. Vocal cords
3. Rear mouth resonance chamber
4. Opening from rear to front resonance chamber
5. Front mouth resonance chamber
6. Opening from mouth

These six are more or less in the direct path of speech, but in a sort of a by-path we have two more:

7. Nasal resonance chamber
8. Position of soft palate and uvula (opening to nasal chamber).

The eight of these in their action are not completely independent of one another. Thus, 3, 4 and 5 act decidedly in unison. Some do not, or at least need not, vary greatly as 6, the mouth opening, which may be kept fixed for the production of all the vowels. Again, the soft palate 8 may open and close the nasal chamber, intermediate positions being unimportant. The eight variables given then, actually may be reduced to five or six in practice.

We come next to the production of the remaining sounds which are classified as fricatives and stop consonants, and are again divided according to whether they are voiced or unvoiced. The voiced ones require the use of the vocal cords; the unvoiced ones do not. As set forth in my application, Serial No. 47,393, above referred to, they comprise eight fricative consonants (four voiced such as v, and four unvoiced such as f) and eight stop consonants (four voiced such as b and four unvoiced such as p).

The fricative consonants are produced with about the same position throughout of the vocal organs except that a certain air outlet or aperture is formed at varying places. Thus for v and f it is formed from the lip to the teeth; for z and s it is formed from the upper teeth to the lower teeth; for the two th sounds it is formed from the tongue to the teeth; for the zh and sh sounds it is formed from the tongue to the hard palate. The voiced consonant is made by pronouncing the unvoiced consonant but vibrating the vocal cords at the same time as though to increase the volume.

The stop consonants are made by forming a stop to the passage of air in the mouth at some particular point, building pressure up behind this and then opening rapidly at the closed point so as to give an explosive sound. The stop is formed by the upper lip against the lower lip in the case of b and p, by the tongue against the upper teeth in the case of d and t, by the tongue against about the middle of the hard palate in the case of j and ch, by the tongue against the soft palate in the case of g and k. In going from the unvoiced to the voiced consonant the formation of the stop, or for that matter of the opening of the outlet in the case of the fricative consonants, may be slightly further front or backward.

In producing the fricative and stop consonants the different parts of the vocal system are used differently than in the production of vowels. Thus, the nasal resonance is of little importance, the vocal cords are not used in producing the unvoiced consonants, the large air chambers in the front and rear mouth are of much less importance, and two new and very important factors are added to the position at which a closure is partly made and held in the case of the fricative consonants and completely made but not held in the case of the stop consonants. To list the independent variables again in the same order as before, we have:

1. Lung pressure
2. Vocal cords (for voiced consonants)
3. Nasal resonance chamber
4. Rear mouth resonance chamber
5. Opening between air chambers of mouth (for fricatives)
6. Front resonance chamber of mouth
7. Position of closure or explosive opening.

Here, as in the case of vowel production, we have more parameters than are essentially independent with any large degree of freedom. Thus the vocal cords are only used for part of the consonants, the nasal resonance is not very important, the size of the mouth resonance chambers is probably of limited importance. The position of the closure or opening is very important, but the two are essentially the same sort of parameter so they are shown as one rather than two. Accordingly, we conclude again, that of these parameters 5 or 6 is ample to represent the actual variable characteristics in speech production.

There are a number of odd effects that in the discussion up to this point have not been allowed for to any extent, at least not intentionally. One thing of this sort is odd deformities or deficiencies in the usual oral structure. Other odd effects that we have are those produced when we do unusual things with the voice such as whisper, talk in a falsetto tone, produce ventriloquistic sounds, or produce what is called double voice.

For all of these odd effects it is probably reasonable to allow two or three degrees of freedom further. However, as the eight original degrees of freedom were considered to be essentially less than eight, it would seem that an allowance of eight for the total might be approximately right. If we desire to be generous, perhaps, we should say ten. In speech over telephone circuits of limited frequency range the number might be 20 per cent or 30 per cent less than that required for high quality speech production; i. e., seven or eight independent variables may suffice for commercial telephone transmission and ten for high quality transmission.

If we vary any of our speech producing variables as rapidly as the controlling muscles permit, we find the limiting speed is about eight or ten times per second. Accordingly, each variable has a fundamental of 10 cycles or less while producing speech.

We are now in position to calculate the band width required for transmitting the speech defining signals. On the basis of transmitting as the useful band not more than 100 per cent of the fundamental frequency (this is done in television and telephotography) we require a frequency band $10\times10\times1.00$ or 100 cycles for transmitting ten speech defining signals. This 100 cycles is more or less of an absolute minimum for high quality speech no matter how much money we are willing to spend. In telegraph more than 100 per cent of the fundamental frequency is required as the minimum transmitted band. In unloaded submarine telegraph cables it is about 110 per cent, while in voice frequency telegraph where frequency space is less important it uses about 350 per cent for either side-band and transmits both. A 200 per cent basis is quite conservative. This would mean 200 cycles of useful frequency band.

Having found that there are approximately ten independent variables in speech production, in setting up a circuit for artificial production of high quality speech we then need ten independent parameters. However, we need not use in any narrow frequency range transmission scheme these same ten. So long as the parameters are entirely independent we know mathematically that we can use any ten we choose. Not only can the ten be chosen in any fashion provided they are independent, but if they are not entirely independent enough more can be chosen to make up for the lack of independence. It is advantageous to pick the ten that from an engineering standpoint give a desirable design or the ten that give an optimum design. A particular case of much interest is that where most of them are the amounts of power in sub-bands of the frequency range of speech.

The interest in this case arises from the fact that it is based on using as parameters those physical quantities that are most easily measured. The easiest thing to measure is power, including current and voltage as measures of power and the easiest way to separate power into parts for the extra variables needed to measure is by dividing it up according to frequency bands. After the power frequency characteristic is measured the sound spectrum to be transmitted is entirely defined except for the power distribution within a frequency sub-band. This last needed factor, in view of the nature of the energy spectrum of speech sounds, is given by the fundamental frequency of the speech sound, considering this frequency to degenerate to zero for the unvoiced sounds. Fortunately we have found means of measuring this fundamental frequency also, thus giving a complete set of specifications for reconstructing the speech sounds. A specific system of this sort is described hereinafter. Any other sufficient set of measurable physical qualities could, of course, be used equally well.

The specific system is shown in a general schematic form in Fig. 1. Speech coming from the line west reaches the west terminal of the system. For simplicity a four-wire connection obtained with a bridge transformer or hybrid coil and a balancing network N, is shown at this point. The transmission may be over a two-wire circuit, the circuit modifications being obvious. The incoming speech is transmitted through circuit 1 to the west speech frequency compressing circuit 2. Fig. 2 is a detailed showing of this west transmitting circuit or speech frequency compressing circuit 2. The frequency range of the speech is considerably decreased in circuit 2 and then the resulting signals are transmitted over the line L of limited frequency range of transmission to the receiving terminal at the east. Here there is a speech frequency restoring circuit 3 which sets up speech signals more or less the reproduction of the speech signals energizing the west sending circuit 2. Fig. 3 is a detailed showing of this east receiving circuit or speech frequency restoring circuit 3. These speech signals set up by circuit 3 are then transmitted on through speech receiving circuit 4 to the line east. Transmission in the opposite direction is carried on over limited frequency line L′ by the same process, the east speech frequency compressing circuit 2′ being like circuit 2 except oppositely pointed, and the west speech frequency restoring circuit 3′ being like circuit 3 except oppositely pointed.

The simplest circuit to use perhaps is an electrical one where the amounts of power in small frequency sub-bands of the speech frequency range are used as independent variables. The power in any one such sub-band is not entirely independent of that in the others but is sufficient so that we do not need many extra sub-bands on this account. Such a circuit is relatively easy to build electrically with filters, modulators, equalizers, etc., as circuit elements.

Using an all-electrical circuit of the type suggested we must get a satisfactory electrical definition of speech signals. The electrical speech signals may be defined as having two characteristics, (1) a frequency pattern and (2) an amplitude vs. frequency pattern. These are the fundamental elements that characterize speech signals, that is the elements that make one sound different from another.

The frequency pattern in speech seems to be of two types. In vowels and near vowels there is a fundamental frequency with a large number of upper harmonics. For unvoiced sibilant consonants there is a more nearly continuous energy spectrum somewhat similar (except in amplitude characteristic) to that of resistance noise. For other sounds there may be a mixture of these two patterns with one or the other predominating. For each frequency pattern there is of course an amplitude-frequency characteristic.

This dual nature of the speech signals as defined electrically leads to a dual type sending or frequency range reducing circuit as shown in Fig. 2 and a dual type receiving or speech frequency restoring circuit as shown in Fig. 3. The speech currents entering the sending circuit **2** in Fig. 2 energize a frequency pattern control circuit FP and an amplitude pattern control circuit AP. The frequency pattern control circuit comprises but one channel and discriminates as to the frequency pattern. This discrimination includes discrimination as to the fundamental frequency when there is one. The amplitude pattern control circuit branches into ten channels AP_1 to AP_{10} and determines what frequency amplitude pattern we have. The information obtained from these two analyzing elements FP and AP will then be expressed in the form of electrical currents whose potentials may be applied to the segments zero to 10, inclusive, of the rotary distributor RD where they may be picked up by the brush of the distributor and applied to the limited frequency line ML.

At the receiving end of the line ML as shown in Fig. 3, the distributor RD', which is synchronized with the distributor RD, picks up the transmitted signals from the line after being amplified by the receiving amplifier RA. The signals are then applied to the segments zero to 10 of the distributor which segments are associated with the frequency pattern control circuit FP' at the receiving end and with the amplitude pattern control channels AP_1' to AP_{10}' of the receiving apparatus. The potential received in the frequency pattern control circuit FP' is applied across resistances B_1 and B_2 to control the frequency pattern sources RN and MV_0 so as to cause current of the proper frequency pattern to flow from these sources in a manner which will be more fully explained later. The potentials applied by the distributor to the amplitude pattern control channels AP_1' to AP_{10}' at the receiving point are used to control shaping networks SN_1 to SN_{10} in the respective channels to give the proper amplitude-frequency pattern to the power received from the energy source RN or from the multivibrator MV_0, as the case may be. We then have reproduction of the original speech signal for any further transmission in the ordinary manner.

The problem arises of selecting frequency bands to use in the amplitude-frequency characteristic determination. A basis for solution of the problem is found in the function known as the importance function, which has been established by articulation studies. This function varies with the amplitude of speech signal received by the ear. We shall assume that the received speech signal is at the optimum power level. Ten or so independent variables are needed for good speech transmission. Ten such bands for the entire speech frequency range should then be as follows:

Band No.	Importance function	Frequencies	ΔF
			Cycles
1	0 to .1	250 to 530 or	280
2	.1 to .2	530 to 780 or	250
3	.2 to .3	780 to 1,100 or	320
4	.3 to .4	1,100 to 1,500 or	400
5	.4 to .5	1,500 to 1,950 or	450
6	.5 to .6	1,950 to 2,350 or	400
7	.6 to .7	2,350 to 2,900 or	550
8	.7 to .8	2,900 to 3,750 or	850
9	.8 to .9	3,750 to 4,950 or	1, 200
10	.9 to 1.0	4,950 to 7,100 or	2, 150

This is not necessarily the best choice of frequencies that can be made. Other considerations enter, such as, for example, including the 0–250 cycle band, making slight changes in the location of the other bands for naturalness, changing the percentage of increase of frequency in a band, the amount of repetition of pattern in different bands, the noticeable distortion due to smallness or largeness of band, etc. However, these considerations would not greatly affect the final choice of frequencies. Accordingly, this set is deemed satisfactory. It is noted that at the most important places the bands are quite narrow in width, 250 cycles, 280 cycles, etc. The three lowest ones are the three smallest and correspond ordinarily to not more than two harmonics each, in the case of voiced signals. The next three probably correspond to about three harmonics in the average case of a male voice. The next one which carries us close to 3,000 cycles might correspond to four harmonics. Above 3,000 cycles they widen out quite considerably. On this frequency basis we have altogether then, eleven parameters for the entire speech range, one for the frequency pattern control and ten for the amplitude pattern control. Similarly, we require a total of eight parameters to go to 2900 cycles, nine to 3800 cycles and ten to 5,000 cycles.

While it is desirable to use sub-bands of different widths, as has just been pointed out, there may be conditions in which it is desirable to have all the bands of equal width. For example, where the sub-bands are of equal width all of the channel apparatus may be made alike even to the general design of filters for selecting the bands. With the sub-bands of equal width a satisfactory reproduction of speech may be obtained especially if the number of sub-bands is increased.

To give a better picture of the proposed circuit it is helpful to indicate frequencies for the various circuit elements. For this purpose let us assume that we will transmit over the main line a 330 cycle total transmission band with other choices such as to tend to give typical results after overcoming typical difficulties. This 330 cycle band is on the basis of 11 channels each passing a 20 cycle band with a 10 cycle waste space between. Ten of these are for amplitude control as already chosen, and the other is for the frequency control pattern. On this basis we shall proceed to discuss first the frequency control circuits, second the amplitude pattern circuits, then the overall circuit, and finally, general engineering considerations.

Frequency pattern control

A particular form of the frequency pattern control circuit is shown in Fig. 2 at the transmitting end and in Fig. 3 at the receiving end. This circuit must perform a number of functions. At the sending end it must analyze the speech signal to determine its characteristics with respect to the frequency pattern, that is, it must determine whether the sound is a voiced sound involving a discrete frequency pattern or whether it is an unvoiced sound involving a continuous frequency pattern. If the pattern is of the former type it will include a fundamental and harmonics thereof, and the fundamental will from time to time vary in pitch so that the harmonics will be raised or lowered in the frequency range as the pitch varies. Consequently, the circuit will also have to determine the pitch at the transmitting end. At the receiving end the frequency pattern control circuit must determine whether the multivibrator source MV_0 is to be set into operation or whether the resistance noise source RN is to be operated, this selection depending, of course, upon whether the analyzed speech sound involved a discrete pattern or a continuous pattern. If the multivibrator source MV_0 is put into operation it must also be controlled by the frequency pattern control circuit to generate the fundamental corresponding to the fundamental in the speech sound together with the necessary harmonics. It must also control the multivibrator to vary the fundamental together with the harmonics in pitch to correspond with the changes in pitch in the speech sound.

The operation of the frequency pattern control circuit in its selection, as between a discrete spectrum and a continuous spectrum, takes advantage of the fact that in vowels and other sounds having a decided fundamental frequency in the range from 80 to 320 cycles there is a high power level, while in sounds like the sibilant consonants where the power is in a continuous spectrum rather than a discrete one the power is much lower. When a high level discrete spectrum is applied the frequency pattern control circuit sends to the limited frequency line ML a current indicating what the fundamental frequency so applied is, not, however, indicating anything about the amplitude of the fundamental frequency in the speech signal. When a low level continuous spectrum speech signal such as that of the syllabic consonant is applied the circuit is not energized. Consequently, no current is transmitted to the line ML and under these conditions the continuous spectrum pattern generated by the source RN is made available at the receiving end.

Referring now especially to Fig. 2, band-pass filter F_0 selects a band from 250 to 500 cycles of the voice signal so as to be sure to include at least two harmonics of the voiced signal if the fundamental is below 250 cycles. In case the fundamental is around 80 cycles and therefore inefficiently transmitted over the telephone line to the point at which the fundamental is determined, this filter enables us to select harmonics of the fundamental, from which the fundamental may be determined by beating the harmonics together. The output of this band-pass filter is fed to a detector D which may be merely some small copper-oxide elements. If the fundamental is below 250 cycles at least two harmonics will be selected and these will beat together in the detector to produce the fundamental frequency in the output of the detector D. Should the fundamental lie within the range of the filter the fundamental only will be selected and there will be no harmonics present and, in this case, the fundamental will pass through the detector D and will appear in the output circuit.

The output from the detector D is next sent through an attenuating network E_1, of the type often termed an equalizer, which has a loss which increases with frequency. This insures that the fundamental frequency will come out of this network at a higher power level than any upper harmonics of the fundamental which may be present in the output of the detector as the result of the detecting operation. For practical purposes this purifies the fundamental tone. Next, the output from this equalizer is fed to a constant output amplifier LA so that from this amplifier there is obtained essentially a single frequency, the fundamental of the speech signal, at a constant power level, regardless of what frequency it is.

This fundamental frequency may be from about 80 to 320 cycles. We next pass the fundamental frequency at this constant power level through another equalizer E_2 similar to the one described previously, so that the output from this equalizer decreases as the frequency increases. This results in changing the amplitude of the fundamental in accordance with its frequency. The output of the equalizer E_2 is then sent through another copper-oxide detector D_0. Essentially this detector produces a direct current bias which fluctuates in amplitude as the fundamental frequency of the speech varies.

This output is now passed through a pad P_0 to an integrating circuit I_0 which may consist of a shunt resistance of 12000 ohms, a series resistance of 10^5 ohms, and a capacity of 2 microfarads. These values may, of course, be varied as the circuit design dictates. The function of the integrator is to integrate the fluctuations of potential over the small interval of time between two successive commutations of the rotary distributor RD so that the distributor will pick up the integrated voltage over this period of time instead of the instantaneous voltage that happens to exist at the moment the brush of the distributor RD passes over the zero commutator segment of the frequency pattern control channel. The pad P_0 is in circuit between the integrator which has a relatively high impedance and the output of the detector D_0 which has a relatively low impedance to introduce an impedance match between these two elements. The potential picked up by the distributor is applied to the main line ML and amplified by the transmitting amplifier TA. After passing over the main line ML the current is amplified by the receiving amplifier RA and its potential is picked up by the brush of the rotary distributor RD' in Fig. 3 and applied to the zero segment of the receiving frequency pattern control circuit FP'.

The operation so far described is what takes place when the voice signal involves a discrete pattern as in the case of voiced speech. Let us now return to the transmitting apparatus of Fig. 2 and see what happens in the frequency pattern control channel FP when the voice signal involves a continuous spectrum. In this case the band-pass filter F_0 selects a continuous range of frequencies between 250 and 500 cycles and, hence, no fundamental appears in the output of the detector, but the entire band of frequencies selected by the filter will all appear there. The equalizer E_1 can not now have the effect of selecting a fundamental to the exclusion of other frequencies although it may favor the lower frequencies in the band. These are next applied to the constant output amplifier and are then applied to the equalizer E_2. It will be remembered that the purpose of this equalizer in the case of a voiced signal was to change the amplitude of the selected fundamental in accordance with its frequency. It cannot, however, have such an effect in the case of the band now being considered, but will merely serve to further discriminate in favor of the lower frequencies of the band. Whatever its action, it will be the same for all unvoiced signals, for the same continuous sub-band will be selected for any unvoiced signal, as the spectrum is continuous. The band of frequencies appearing in the output of the equalizer will now be applied to the detector D_0 which will produce in its output essentially a direct current. This direct current, however, will be of very small amplitude as compared with the corresponding current resulting from the detected fundamental frequency in the case of a voiced sound, and this fact is taken advantage of in selecting the comparatively high frequency source at the receiver. The small direct current appearing on the output of the detector D_0 is transmitted over the line to the receiving frequency pattern control channel FP′ in Fig. 3, as described.

Now coming to the receiving arrangement in Fig. 3, the potential applied to the zero segment of the distributor RD′ is applied to a filter F_{30}'. This is a low-pass filter passing a band from zero to about 20 cycles and may be identical with the filters F_{31}' to F_{40}' of the frequency amplitude control channels. The current applied to the filter F_{30}' is a momentary pulse which exists only during the time the wiper passes over the segment zero, and the filter acts to hold over or restore the impulse during the entire time interval which exists until the brush again passes over the zero segment.

It will be remembered that the frequency pattern control current transmitted over the line was a substantially zero current in the case of a continuous spectrum, but in the case of a discrete spectrum it was of considerable amplitude, and this considerable amplitude varied in accordance with the frequency of the fundamental of the voiced sound. Considering the latter case first, the fluctuating direct current passing through the filter F_{30}' serves two purposes.

First, it effectively disables an amplifier VA in Fig. 3 which would otherwise amplify resistance noise received from the resistance R through amplifier A. The biasing current from the output of the filter is so applied to a grid biasing resistor B_1 for the amplifier that when substantially no bias is received (as is substantially the case for a continuous spectrum) the resistance noise from R and A is passed on through the amplifier VA. However, when substantial bias is received, as is the case with a discrete spectrum, the gain of the amplifier VA is decreased by a negative bias being applied, so that substantially no resistance noise is transmitted.

Second, the current from the filter F_{30}' is applied to a biasing resistance B_2 in the common grid lead of a push-pull vacuum tube circuit VR. The grid circuits of the two tubes of the amplifier VR, it will be noted, are connected in parallel, but the plates are in series. The purpose here is to control the plate resistances of these tubes by the biasing current. The plate resistances in series are used as the resistance element R_0 of a multivibrator circuit MV_0 so that the frequency of the multivibrator circuit is controlled by this variable plate resistance R_0. It is controlled in such a way as to set up the desired fundamental frequency of voice plus all of its harmonics. To insure both even and odd harmonics, the circuit is arranged to take off the outputs from the two tubes of the multivibrator in series and in parallel and then combine these two so as to generate all the harmonic frequencies. Another possible arrangement of the multivibrator is to have it designed so as to generate one-half the fundamental frequency from which only the even harmonics are used. With the arrangement as shown, however, the fundamental frequency generated and the harmonics thereof will vary in frequency in accordance with the amplitude of the biasing current which in turn varies in accordance with the frequency of the fundamental in the voiced signal.

The foregoing applies to the case where the spectrum is discrete. Now let us take up the case of a continuous spectrum. When the signal involves a continuous spectrum no bias current or at least substantially no bias current is present. Under these conditions, the multivibrator circuit MV_0 stops oscillating, and as the amplifier VA is unbiased at the resistor B_1 the resistance noise is amplified and transmitted.

The multivibrator output and the resistance noise circuit output from the variable gain amplifier VA are combined in the circuit leading to the amplitude controlling circuits through filters F_1' to F_{10}', inclusive. Preferably the mutivibrator output is first passed through an equalizer E_4 which serves to make the output power the same for each frequency, fundamental and upper harmonics. If desired, this end can be obtained by making the coupling loose between the primary and secondary windings of the multivibrator output transformers, the equalizer E_4, in this case, being omitted.

Returning to Fig. 2, the attenuation-frequency characteristic of equalizer E_2 determines the variation with frequency, of the amplitude of the fundamental components of voiced sounds that are delivered to detector D_0. This input to detector D_0 has its syllabic frequency component detected by detector D_0 and transmitted to the bias resistor B_2 in Fig. 3, which determines the fundamental frequency of the multivibrator MV_0. Consequently, the voltage output of equalizer E_2, at each fundamental frequency that E_2 transmits, should be of such value that the voltage across the resistor B_2 in Fig. 3 will have the proper value to cause the multivibrator to assume that fundamental frequency as the fundamental frequency of the multivibrator. Therefore, if the voltage transmission-frequency characteristic of the system from the output of equalizer E_2 to the bias resistor B_2 is, for example, flat, or in other words, if the ratio of the equalizer output voltage to the resulting voltage across the resistor B_2 is independent of frequency, then the attenuation-frequency characteristic of the equalizer E_2 should be such that the ratio of output voltage to input voltage for the equalizer decreases with frequency increase in the same manner that the fundamental frequency of multivibrator MV_0 increases with decrease of the biasing voltage across the biasing resistor B_2. Then, the fundamental frequency set up by MV_0 will increase and decrease in the same manner as the fundamental frequency of speech sound waves in the input of equalizer E_2.

The result of all this is that a frequency pattern will be applied to the common circuit leading to the filters F_1' to F_{10}', inclusive, at the receiving point, which will be continuous, and extend over the entire voice range from 250 to 7100 cycles in the case of an unvoiced sound at the transmitter, whereas in the case of a voiced sound the frequency pattern applied to the common circuit of these filters will be a discrete frequency pattern having a fundamental and its harmonics with the fundamental varying up and down in accordance with the pitch of the voiced sound.

Amplitude pattern control

The next matter to be considered is how the frequency patterns thus generated at the receiving point are to be controlled and modulated to reproduce speech, for it will be clear that unless they are modulated in some manner we will merely hear a resistance noise sound somewhat like the roar of the surf at the seaside in case the resistance source is active, and in case the multivibrator source is active we will merely hear a sound somewhat like that of an ordinary buzzer. In order to modulate these sounds to produce speech, therefore, the amplitude pattern control circuits are provided and, as herein shown, they are ten in number.

The amplitude pattern control circuits are essentially circuits which at the transmitting end measure how much power there is in the speech signal in a suitable number of chosen small frequency bands, and this information is transmitted by control currents to the receiving end where the output of resistance noise from VA or multivibrator harmonics from the multivibrator MV_0 is shaped accordingly. These frequency bands are chosen as described previously.

The transmitting portion of the amplitude pattern control circuits is shown in Fig. 2 and the corresponding receiving portion in Fig. 3. The channels, AP_1 to AP_{10} and AP_1' to AP_{10}', are used to transmit information about the amplitude pattern. Thus, the channel corresponding to segment 1 of the distributor RD transmits information about the amplitudes in the speech range from 250 to 530 cycles, and the channel corresponding to segment 2 transmits information about the range from 530 to 780 cycles, and so forth.

Considering the channels AP_1 and AP_1', for example, the output from the 250–530 cycle speech band-pass filter F_1 is fed to a detector D_1, which may be, for instance, of the copper-oxide type. The syllabic frequencies in the output from this detector vary in amplitude as the portion of the speech signal selected by the filter F_1 varies in amplitude and pass through the pad P_1 and the integrator I_1 to segment 1 of the rotary distributor RD. As in the case of the integrator I_0, the integrator I_1 serves to integrate the potential variation over the period during which the distributor passes from the segment 1 over the other segments and back again to segment 1. The pad P_1, of course, serves to establish the proper impedance match between the integrator and the detector D_1. The potential applied to segment 1 is picked up by the sending brush S of the distributor and applied to the sending bus-bar SB from which it is transmitted over the main line ML to the receiving bus-bar SB'. Just after the brush S at the transmitter passes from the segment 1 a ground brush G passes over the segment and discharges any remnant charge in the condenser of the integrator to ground over the ground bus-bar GB. The grounding action should be as short as possible in order to permit the succeeding element of the wave to start charging the condenser corresponding to C in the integrator as soon as possible. Ground bus-bar GB may therefore be made up of a number of very narrow segments tied together to ground. This restricts the time of contact between the brush and the ground connection to a very short interval.

The signal picked up by the segment 1 of the receiving distributor RD' is applied to the receiving amplitude pattern channel AP_1' passing through the filter F_{31}' which acts like the filter F_{30}', as previously described, to hold the signal over during the period required for the receiving brush S' to make one complete revolution and come back again to the segment 1. The variable direct current passing through the filter F_{31}' is applied to a biasing resistor B_3 to give a grid bias to a signal shaping network or push-pull amplifier SN_1, which bias will vary in accordance with the power of that portion of the speech band which passed through the filter F_1 at the sending end. The amplifier SN_1, consequently, amplifies that portion of the frequency pattern generated by the multivibrator MV_0 or by the resistance noise source VA which is selected by the 250–530 cycle speech band-pass filter F_1'. The modulated output is then fed through a 250–530 cycle speech band-pass filter F_1'' to the input of the speech amplifier SA, where the outputs from nine other speech band-pass filters (of channels AP_2' to AP_{10}') are combined to give the original speech signal. The speech currents are then transmitted through amplifier SA to the speech receiving circuit 4.

It will be understood that the transmitting channels AP_2 to AP_{10} at the transmitter are like channel AP_1 except as to frequencies involved, and receiving channels AP_2' to AP_{10}' at the receiver are like channel AP_1' except as to frequencies involved. Likewise, the detectors D_0 to D_{10} may be alike. For most cases the integrating circuits I_1 to I_{10} and the pads P_1 to P_{10} may be alike. So also, the variable gain amplifiers or signal shaping networks SN_1 to SN_{10} may be alike and filters F_{30}' to F_{40}' may be alike.

Overall circuit considerations

A number of non-linear circuits have been shown. One type, D_0 to D_{10}, is referred to as the detector circuit and, in general, its purpose is similar to that of detectors in voiced operated circuits, that is, the received speech power is more or less rectified giving a current component of syllabic frequency to be used as a control. The detector D, for obtaining the fundamental frequencies of speech sounds, is similar. The constant output amplifier LA is also a non-linear circuit.

At the receiving end the control currents are used for two purposes, each of which involves a type of non-linear circuit, (*a*) to bias amplifiers VA and SN_1 to SN_{10} and thus change the gain through them, and (*b*) to vary the plate resistance of vacuum tubes in circuit VR. Also, at the receiving end of the circuit there is the multivibrator or relaxation oscillator MV_0 for setting up the frequencies in the vowel sounds.

In the circuit design it is important to see that the delay in all the different branches is the same. Since the frequency pattern control branch tends to have more inherent delay than the amplitude control branches, it is desirable to have a certain amount of delay in common with all the amplitude control circuits as is indicated by delay equalizer DE in Fig. 2.

A set of band-pass filters F_0 to F_{10} are provided at the transmitting end, and a set of filters F_1'' to F_{10}'' are provided at the receiving end. If desired, a third set F_1' to F_{10}' may be used at the receiving end making twenty instead of ten speech band filters in the receiving amplitude control circuits, for the case of the 7100 cycle band to be transmitted. These ten or twenty speech band filters at the transmitting and receiving ends, respectively, might have roughly, requirements of about 3 decibel loss discrimination at the separating points between the two bands, 20 decibel loss at the middle of the next band and 40 decibel loss at the middle of the second band. Assuming the requirements to be approximately these, the delay in the speech band filters is relatively small, two of them in tandem giving only about 10 milliseconds delay. These speech band filters can probably have even easier attenuation requirements than indicated above and the delay may therefore be made even less.

If desired, to simplify delay equalization in the system, the speech band-pass filters may be made of constant width as, for example, 300 cycles. For instance, for transmitting 250 to 2950 cycle speech, nine amplitude pattern control channels have been used, with filters F_1 to F_9, respectively, passing the speech bands 250–550 cycles, 550–850 cycles, 850–1150 cycles, 1150–1450 cycles, 1450–1750 cycles, 1750–2050 cycles, 2050–2350 cycles, 2350–2650 cycles, and 2650–2950 cycles. In this case filters F_1' to F_9', respectively, were arranged to pass these same bands and, likewise, filters F_1'' to F_9''.

Another simplification of the circuit of some interest may be obtained by eliminating the frequency pattern control channel FP and FP'. We then cannot switch from the multivibrator circuit MV_0 to the resistance noise source FPS, but must use either the resistance noise or the multivibrator now set at an arbitrarily chosen frequency such as the average vocal cord frequency. Interesting sound effects are thus produced. With the fixed frequency multivibrator source the manufactured speech has the sound of a whisper. Both circuits, however, give good intelligibility at a considerable reduction in the circuit equipment required and may, therefore, be useful where naturalness is not an important factor or not a requirement in transmitting speech sounds. Another possibility, of course, is to vary the pitch of the multivibrator source by means of dial. In this way the emotional character may be introduced into the artificial speech, or it may be given some abnormal or bizarre effect.

Ordinarily, with a system of the type shown, the permissible range of volume transmitted might well be somewhere from 20 to 40 decibels, for example. The wider the frequency band the easier it is to get a large number of volume range steps at the receiving end without too fine adjustments of the apparatus. The smaller the volume range, the more readily can transmission be carried on over long circuits of high attenuation such as submarine cables. That is, signal levels down much nearer to resistance noise can be used and therefore the cable can be a cheaper cable of greater attenuation than could otherwise be used, or a wider frequency range can be used on a given cable.

Any desired portion of the circuit or link between the terminals of the system may be a two-wire, four-wire or radio circuit. It will be understood that for multiplex carrier transmission of a plurality of conversations simultaneously over a common transmission circuit or medium, either by wire or by radio transmission, each speech channel may have its speech band reduced in frequency range by a speech frequency compressing circuit (such as the circuit **2** of Figs. 1 and 2) and then applied to the modulating channel of the wire carrier or radio carrier multiplexing apparatus at one end of the system. At the other end the band detected by the demodulator of the wire carrier or radio carrier multiplexing apparatus will correspond to this narrowed band and may be impressed upon a speech frequency restoring circuit (such as the circuit **3** of Figs. 1 and 3).

It is emphasized that the output of the speech frequency compressing circuit **2** of Figs. 1 and 2 is unintelligible, rendering it difficult for unauthorized persons to tap the circuit and affording a high degree of secrecy in either wire or radio transmission. The waves transmitted, through the medium L, directly contain no intelligibility whatever.

Where the circuit connecting the terminals of the transmission system is, for example, a trans-Atlantic submarine telephone cable, the reduction of the frequency range transmitted can be of great aid especially because of the difficulty of inserting the desired number of amplifiers to keep down the attenuation at high frequencies.

In multiplex telephony over cable or open wire lines or by radio transmission, the reduction of the frequency range can facilitate an increase of the number of channels on a line or in the transmission medium, as for example, the placing of two or more speech channels, on the line or in the transmission medium, in the frequency range ordinarily occupied by one.

The compression of the frequency range of the transmission, in accordance with the invention, can increase the signal-to-noise ratio obtaining at the receiver without necessitating increase of the sending level of the signal. Thus if the frequency band is reduced by a factor of 10 and the same sending level used there is a gain of 10 decibels in the signal-to-noise ratio because the noise power is proportional to frequency band. This is of especial importance in long radio transmission systems such as the trans-Atlantic, for example, in which the transmission medium is unsatisfactory for the desired grade of transmission of the original communication signals because of inadequate signal-to-noise ratio in the transmission medium.

The compression of the frequency range also has the advantage that for a radio system the chances of fading in a given band are reduced. Furthermore, for a given available frequency range, it is possible to divide the range into a greater number of channels, each of reduced range, and certain channels may then be set aside as spares to be used when any of the regular channels fail through fading.

General engineering considerations

As will be pointed out later, a study of the commutator effect reveals that it acts like an aperture in television and therefore functions as a low-pass filter (see pages 601 to 603, Bell System Technical Journal of 1927), as will be discussed in more detail later. The commutator has the advantage, over the low frequency carrier method of transmitting the control signals as disclosed in my application, Serial No. 47,393, that it involves no delay or delay distortion in the usual sense. By using the commutator at both ends of the restricted frequency line the carrier band selecting filters at both terminals of the system are eliminated. The time between two successive rotations of the distributor is not true delay because the two successive commutations of a particular channel current wave that occur here involve merely the transmitting of the amplitude values of the current waves from interval to interval along the length of the wave as the wave progresses. There is, of course, the action of the integrating circuit in which the charge between two successive commutations along the length of the wave is held between commutator brush wipings, and this gives an average delay equal to one-half of this period, but this is merely a storage delay.

In order to clarify the action of the problem, graphs have been drawn, as shown in Fig. 4, representing the wave forms in the frequency pattern control channel and in the ten amplitude pattern control channels. The waves represented in each channel in Fig. 4 are the detected or rectified waves appearing in the output of the detectors D_0 to D_{10}, respectively, of Fig. 2, if the high voice frequencies are cut out and only the low syllabic frequencies retained. It will be noted that each of these wave forms is repeated at successive intervals, thus representing the repetition of the same syllable. The graph is presented in this form for the reason that it is easier to picture what happens if the same sound is repeated over and over to give rise to steady-state features, for after all, the system that transmits a repeated sound well will transmit the same sound well if it is given but once with other sounds substituted before and after. As will be seen from the wave depicted in the zero channel, the frequency pattern control current will, in general, be a more or less steady current with pulses corresponding to the syllabic frequencies of speech. The waves drawn for the amplitude pattern control channels are not drawn to represent any particular sound but are more or less typical values such as might reasonably be expected to occur.

Repetition of the syllables is assumed at the rate of five per second because it has been found that simple sounds can be made at the rate of six or seven per second, ("tut" 6.7 for instance) while more complicated sounds like "start" can only be enunciated at the rate of about three per second. It is therefore concluded that five sounds per second or three hundred per minute is a fair rate to assume. This time allowance is considered quite ample, for "tut" is really two sounds and therefore involves enunciation at the rate of twelve or fourteen per second, while "start" is really five sounds involving enunciation at the rate of fifteen per second as against not over five per second observed on ordinary telephone conversations.

The amplitude characteristics of the currents in the several frequency band channels or amplitude pattern channels are shown different from channel to channel, for it has been found to be so in oscillograms taken for actual cases. It should also be noted that the currents in the different channels are assumed to be integrated over intervals of 40 milliseconds or $\frac{1}{25}$ of a second (the time assumed for one rotation in the distributor) so that the integrated current between successive commutations, as represented by the dotted lines, is an average of the various current values occurring between two successive commutations. This integration represents the effect of the resistance condenser circuits of the integrators I_0 to I_{10} of Fig. 2.

We now have the following data:

Line frequency=25 per second. (This line frequency corresponds to the term line frequency of scanning as used in television. It is the rate of scanning from element to element along the length of a given wave as distinguished from the scanning from channel to channel along the frequency spectrum.)

Number of elements per line=11

Number of elements per second=25×11=275

$$\text{Element scanning frequency} = \frac{275}{2} = 137.5 \text{ cycles}$$

Required line frequency=137.5 cycles×1.1 to 1.4 =151 cycles to 192.5 cycles.

The element scanning frequency is effective in transmitting channel to channel variation. These variations are, of course, plotted one above the other in a vertical line in the diagram of Fig. 4. If we take the dashed lines in the third column, for example, and plot them horizontally instead of vertically, we will get a curve such as curve *c* of Fig. 6, representing the current variation from channel to channel. This curve is plotted as a smooth curve drawn through the individual amplitude values represented by the dashed lines in column 3 of Fig. 4. This gives the third vertical scanning line of Fig. 4 to be transmitted in the time interval from 80 to 120 milliseconds. The first vertical column will be transmitted in the time interval from 0 to 40 milliseconds, the second vertical column in the interval from 40 to 80 milliseconds, and so forth.

In order to understand what is actually transmitted over the transmission line as the result of the commutator action, each sound may be thought of as having a characteristic amplitude varying with frequency, as shown by curve *c* of Fig. 5, in which ordinates from 1 to 10 correspond to different channels or different frequency subbands. This energy is not fed out as uniform power but is a power that increases from zero up to its maximum value as represented by the curve *c*. This corresponds to shifting the curve up and down as represented by the curves *a*, *b*, *c*, *d* and *e* in the order stated, these curves representing the conditions for the five intervals of time shown in Fig. 4 between zero and 200 milliseconds, the assumed duration of a syllable.

The curves of Fig. 5 represent the assumed conditions resulting from amplitude shifts alone. There is, however, an effective wave distortion which must be taken into account, that is, the change of the shape of any one of the curves *a*, *b*, *c*, *d* and *e* during the progress of the sound. This effect may, for instance, be a shift of resonance from an initial position at lower frequencies to a final position at higher frequencies. This effect is plotted in the case of the curves *a*, *b*, *c*, *d* and *e* in Fig. 6. Comparing these curves with the corresponding curves of Fig. 5, it will be seen that the curve *a* is increased in amplitude over the range from 1 to 4 which may be thought of as corresponding to a degree of resonance at point *v*. Curve *b* in Fig. 6 is drawn to show resonance at the point *w*. Curve *c* shows resonance at point *x*. Curve *d* shows resonance at point *y*, and curve *e* shows resonance at point *z*. By comparing the individual values of these curves from channel to channel with the integrated values represented in the dotted lines in the first five time intervals of Fig. 4, it will be seen that there is an exact correspondence. While these resonance and other similar effects may take place as indicated at syllabic rates, it should be understood that the curves are not drawn from actual findings from an actual study of oscillographs of speech sounds, but are merely typical curves. It is known that an average speech sound corresponds to a shifting of a multiplicity of resonant ranges in some cases at least.

As actually drawn, the curves of Fig. 4 assume that scanning takes place along the vertical division lines at the time intervals indicated at the bottom of the sketch. In practice, this could only be done by introducing successively graded delay networks in each channel so that as the brush rotates from segment to segment it would encounter the wave of each particular channel at corresponding intervals of time along the progress of each wave. As such a complicated storing arrangement would be impractical from a cost standpoint, the actual scanning from channel to channel would take place as represented by the dashed diagonal line shown in the first vertical column of Fig. 4. Since the wave is integrated over the 40 millisecond time interval the slight deviation from vertical scanning, actually involved, is of small consequence.

Amplitude changes from channel to channel in the vertical direction correspond to the variation in amplitude of the signal with frequency, and are well transmitted. If we consider the effect in the direction at right angles to the direction of scanning, we find a different condition. This direction gives the amplitude-time characteristic of the wave in each channel, whereas the vertical scanning from channel to channel gives the amplitude-frequency characteristic of the signal at a given time interval. In .040 second the commutator brush must contact the frequency pattern segment and ten amplitude pattern segments. This allows somewhat less than .004 second for each segment. This time may be spent on the commutator section or any fractional part thereof.

During this time the commutator segment may have either of two things applied to it: (1) the instantaneous voltage picked off a properly terminated circuit, or (2) the voltage (or current resulting therefrom) integrated by a condenser-resistance circuit. On the other hand, a low-pass filter may be inserted after the rectifier to pass the syllabic frequency pulses of the order of 10 cycles or less and cut out the higher telephonic frequencies, or, such a filter may be omitted.

Considering these possibilities together, it seems preferable to use an integrating circuit rather than pick off instantaneous voltages for, in this case, the low-pass filter becomes unnecessary. In integrating, the higher telephonic frequencies will alternate many times per second during a commutating interval with the result that many complete oscillations at the higher frequency rate will be applied to the integrating circuit. In such case the effect of the integrator on the high frequency oscillations is to cause the positive half of the wave to cancel out the negative half, leaving no net high frequency change unless the last oscillation does not go through a complete cycle just before the end of the integrating period. In this case the integrated result will be small as it is, in effect, distributed over the entire integrating period. This corresponds to a very appreciable loss at the higher frequencies of the wave.

For example, consider the lowest frequency band of speech selected, which is from 250 to 530 cycles. Integration occurs over one twenty-fifth of a second and includes ten complete cycles at 250 cycles per second. At this frequency the positive half-cycles of the wave are all canceled by the negative half-cycles, giving an infinite loss. The loss decreases from 250 cycles down to a minimum loss at a half period point which is one-half of twenty-five cycles or twelve and one-half cycles away as shown in the curve of Fig. 14. Thus at 262.5 cycles per second there will be a summation of ten and a half cycles during one twenty-fifth of a second. The first ten of these cycles are canceled out, the positive half against the negative half. The remaining half-cycle is averaged over a period corresponding to ten and one-half cycles which is 21 times longer than the time of the last half-cycle. This amounts to a loss of about 26½ decibels, and would be more than ample to suppress the ripple due to this high frequency component being added on to the syllabic frequencies present in the channel. It is therefore evident that the filter action produced by the combination of the commutator with the integrating circuit effectively suppresses the high frequency components without the use of any filter.

With respect to the syllabic frequencies to be transmitted, in scanning along the wave in each channel from point to point with successive rotations of the brush of the distributor (i. e., in the horizontal direction in Fig. 4), the transmission is not always so satisfactory as is the case in scanning from channel to channel. This effect may be described as "penalized" scanning. That the scanning is of this type along the length of the channel wave is due to the fact that we stay on the curve a brief time (less than .004 second) and then leave it for somewhat more than .036 second before we return.

A better understanding of this may be obtained if we consider a single frequency component having a half-period equal to the time between successive scannings of a particular commutator segment, or the scanning of one vertical line. In other words, if F_c is the frequency of a wave which just completes a full cycle in the time between successive scannings of a given segment, the wave frequency we are to consider will be

$$\frac{F_c}{2}$$

The effect will be seen from the curves of Fig. 10. The upper curve shows the case where the integrating period starts at the zero amplitude point of the wave. In this case we first get a strong positive summation for the first half-wave followed by an equally strong negative summation corresponding to the second half of the wave. As indicated at the right and as represented by the dotted line curve, the only effect has been to put a square top on the wave. If, however, we commence integrating at the 90 degree point of the wave, as indicated by the lower curve of Fig. 10, the first integration gives zero because the positive quarter cycle of the wave cancels the succeeding negative quarter cycle. Similarly, in the second and all succeeding integrations, the positive and negative wave elements cancel each other. The result is equivalent to an infinite loss at this frequency. Obviously, the transmission may be good or bad for scanning in the direction along the wave, depending on the frequency and its phase relation with respect to the scanning interval. It is evident, therefore, that intrachannel scanning is of the penalized type while the interchannel scanning is of the non-penalized type.

The frequency where this trouble due to penalized scanning comes in is that at which the half-cycle wave is completed in one twenty-fifth of a second or .040 second, so that the whole wave is completed in .080 second. In order to understand this more fully let us consider the effect of the commutator and integrating circuit as a filter. Referring to Fig. 15, let us suppose that in a given channel a frequency of 25 cycles per second is being transmitted. Regardless of the point along the particular wave-length at which the scanning operation takes place, it will be evident that during one rotation of the distributor one complete cycle will be integrated. As the positive half of the wave will just cancel the negative half of the wave in the integration, no transmission will take place, and this corresponds to an infinite loss at 25 cycles as shown by the curve in Fig. 15. At any frequency less than 25 cycles, the integration will involve less than one complete cycle so that a complete cancelling out will not take place and some transmission will result as represented by the full line curve from zero to 25 cycles in Fig. 15. Obviously, this curve is similar to that of a low-pass filter in which good transmission can take place to somewhere in the neighborhood of 12½ cycles at which point the loss will only be 3.8 decibels. For frequencies above this, the loss increases very rapidly, as shown by the curve, and transmission will be poor.

At 50 cycles two complete cycles will occur during one revolution of the distributor, and again, the integration would result in a complete cancelling out of the positive and negative halves of the waves with an infinite loss occurring at this frequency, as shown by the curve. At 37½ cycles one and one-half complete cycles would be integrated during one rotation of the distributor with the result that two half-cycles would be canceled out but one half-cycle would be integrated over the entire integration period. This would amount to a loss of about 13 decibels and would be the minimum loss that would occur between 25 and 50 cycles. For frequencies within this range on either side of 37½ cycles the loss would be greater, as indicated by the curve. If now we plot the curve to a smaller frequency scale, as shown in Fig. 14, we will have another minimum loss at 62½ cycles corresponding to about 18 decibels, another one at 87½ cycles corresponding to about 20 decibels, still another at 112½ cycles corresponding to about 22 decibels, etc. Also, there will be infinite loss points at every multiple of 25 cycles. This represents the most favorable condition with no penalizing effect taking place, and it will be evident, therefore, from the curve of Fig. 14, that the system will efficiently transmit frequencies up to 12½ cycles per second, or a little more. But, for any frequency substantially above this, there will be very considerable loss so that the system acts as a good filter.

Coming back again to the curve of Fig. 15, let us consider the effect of penalized scanning. The best condition is that at which commutation takes place at the beginning of the cycle. The worst condition exists when commutation occurs at 90 degrees or 270 degrees later. For a frequency of 12½ cycles one half-cycle is equal to the integrating period and if the commutation takes place at the 90 degree point of the curve, the result will be as indicated by the lower curve of Fig. 10 so that an infinite loss will occur at this point. For frequencies lower than 12½ cycles per second the loss will be less, as indicated by the dotted line in the curve of Fig. 15. For example, consider a frequency of one-half this value of 6.25 cycles per second. Integration under these conditions occurs over one-quarter of the wave-length so that even if commutation occurs at the 45 degree point, some transmission will take place as indicated at the right of the lower curve of Fig. 12 where it shows that a flat-topped wave results having one pulse for each half of the original wave. The scanning here is partially penalized but represents a loss of only about 4 decibels. If commutation occurs at either zero or 90 degrees, two flat-topped pulses occur for each half of the wave as shown at the right of the upper curve in Fig. 12. From the foregoing it will be evident that under the condition of worst penalized scanning we have a loss gradually increasing from zero to infinity over the frequency range from zero up to 12½ cycles as indicated by the dotted line curve in Fig. 15.

Now let us consider a frequency of 25 cycles per second. Here one complete cycle occurs for each scanning rotation and, regardless of the point along the curve at which the scanning takes place, the integration will result in a complete balancing out of the two halves of the wave as indicated in Fig. 8, and we will have an infinite loss. The curve up to 25 cycles, under the worst penalized conditions, will be represented by the complete dotted line curve of Fig. 15

The foregoing discussion applies to the conditions with respect to a given sub-band or channel. When we consider the condition in the main line we have the combined effects of eleven different channels. If the individual channel is good up to 12½ cycles for unpenalized scanning but has an infinite loss at 25 cycles then, as regards the main line, we will have good transmission for frequencies of eleven times these values so that there will be good transmission in the main line up to 137½ cycles with an infinite loss at 275 cycles. For penalized scanning we will have infinite losses at 137½ cycles sometimes and at 275 cycles the loss will always be infinite. Below 137½ cycles for the worst penalized conditions the loss will fall off as shown by the dotted line curve. The circuit will give satisfactory transmission under non-penalized scanning conditions up to 137½ cycles, but for the worst penalized conditions the circuit will only be good up to about one-half of this frequency so that there is a waste of about 50 per cent for this type of scanning. In this respect it is worse than carrier. However, when we consider along with this the virtue of the circuit under non-penalized conditions where it is 100 per cent efficient, we have an average efficiency of about 75 percent, which is about the same as a well designed carrier system.

It is noted that the frequencies mentioned as being well transmitted are only those going to about one-half of the infinite loss frequency in the case of non-penalized scanning and going to one-quarter of that frequency where the worst phasing gives infinite loss in the case of the penalized band. While the attenuation loss is getting high under the latter condition, it nevertheless may be equalized, and even unequalized corresponds only to a 3.8 decibel loss at this point for a single integrating circuit. Later on an arrangement will be discussed for overcoming, in part, the difficulties due to penalized scanning.

Let us now consider the design of the integrating circuit. Referring to Fig. 2, it will be seen that in any channel as, for example, channel FP, the circuit is terminated beyond the pad in the impedance here shown as 12,000 ohms, and beyond this termination is a series resistance R with a shunt capacity C. The voltage across C accumulates until the collecting brush S contacts it and sends the energy over the signal bus-bar SB to the main line. Immediately after the brush S leaves the segment the latter may be connected to ground by means of the grounding brush G to wipe off any residual charge before the next charge is accumulated.

The charging circuit of R and C in series must be designed so that a voltage is about equally effective in charging C whether it comes in the early or late part of the .040 second charging period and regardless of the values of voltage in the rest of this period. This condition will prevail if a fixed voltage applied during the full period of .040 second keeps on charging at a constant rate. This in turn will be the case if the final voltage V_2 on condenser C is not too large a fraction of the voltage V_1 applied to R and C in series. A reasonable value of this ratio may be ¼. Then

$$\frac{V_2}{V_1}=\frac{1}{4}=1-\epsilon^{-\frac{.040}{RC}} \qquad (1)$$

in which .040 is the charging interval in seconds.

Solving Equation (1) for RC we get a value of .17. Thus C might be 1 microfarad and R, 170,000 or any other pair of factors could be used. The larger C the more energy stored and therefore the less loss. The Equation (1) was written to show the method of design for R and C and the order of magnitude of their values but not at all as any limitation on the wide range of values that may be satisfactorily used.

In the circuit arrangement as shown in Fig. 2 an integrating circuit has been provided for a number of reasons, among which is that it has certain advantages in connection with the distributor in performing a selective operation. It is, of course, quite possible to operate the circuit without an integrating arrangement and it is a matter of some interest to compare the action of an integrating circuit with a non-integrating circuit. Let us assume first that the wave-length is just equal to the scanning period as we scan intrachannel along the length of the wave. Then for each successive rotation of the distributor the brush arm will contact the distributor segment of the channel once for each complete cycle of the applied wave.

Fig. 8 shows the result with an integrating circuit. The upper curve represents the case where the commutation occurs at the beginning of the wave cycle and the lower curve indicates the case where the commutation occurs 90 degrees later. In both instances the integration results in a complete cancellation of the wave so that no output current results.

Fig. 9 shows the result for the same wave with a non-integrating circuit, that is, a circuit in which the brush of the commutator picks up the instantaneous voltage of the wave at the particular instant of commutation. In case the wave is commutated at zero nothing will be transmitted to the line. The upper curve shows a case where the commutation occurs 90 degrees later. Here we get a positive impulse of current once every cycle, but the indication is all in one direction. At 45 degrees transmission would also take place, but here again, the pulses of current would all be in the same direction and they would be smaller. If the pick-up occurs under any phase condition other than at the beginning of the wave, current will be transmitted, but the current impulses will be either all positive or all negative. It obviously is useless to try to represent a cyclic wave by what amounts to a continuous direct current. In short, with the instantaneous arrangement we are no better off than with the integrated circuit because while we do get some current the current does not give any information as to the nature of the wave.

Let us next consider a wave of half the frequency of the previous wave or, in other words, a wave whose complete cycle is equal in length to two commutation periods. Fig. 10 shows that with an integrating circuit we will get a positive impulse for each positive half of the wave and a negative impulse for each negative half of the wave if the commutation occurs at the beginning of the wave. This is, of course, satisfactory transmission. If, however, the commutation occurs 90 degrees later, no current will be transmitted. In other words, the scanning will be penalized.

Fig. 11 shows the situation with a non-integrating circuit where the pick-up is instantaneous. Here, if the pick-up occurs at the beginning of the wave no current is transmitted. If it occurs 90 degrees later we get a positive impulse for each positive half of the wave and a negative impulse for each negative half. In this case we also have penalized scanning, but the penalty is applied under different phase conditions than in the case of the integrating circuit.

Finally, let us consider a wave which completes one cycle in four scanning periods or, in other words, a wave of half the frequency of the preceding wave. With an integrating circuit Fig. 12 shows that if the scanning occurs at the beginning of the wave we will get two integrated pulses to represent the positive half of the wave and two integrated pulses to represent the negative half of the wave. On the other hand, if the pick-up occurs 45 degrees later we get an integrated positive pulse representing the middle portion of the positive half of the wave and a negative impulse representing the middle portion of the negative half of the wave. During the intervening scanning, however, there will be no transmission as the remaining positive portion of the wave just cancels the first part of the negative half of the wave.

Fig. 13 shows the condition for instantaneous scanning. Here, if the pick-up occurs at the beginning of the wave we get nothing for the first scanning operation, a positive indication during the second scanning operation, nothing during the third scanning operation, and a negative indication during the fourth scanning operation, etc. If the pick-up occurs 45 degrees later we get positive impulses for the first two scanning operations and negative impulses for the next two scanning operations.

Thus we see that for frequencies up to

$$\frac{F_c}{2}$$

both integrating and non-integrating scanning give substantially the same results, while for frequencies above

$$\frac{F_c}{2}$$

such as for example, the frequency F_c, we get no useful result in either case. The instantaneous method of scanning is, of course, less efficient on any energy basis, than the integrating method.

While this matter of penalized scanning is a serious one, as already pointed out, it is not so serious that the efficiency of the circuit is less than that of a well designed carrier circuit. Nevertheless, we are not without a remedy for the condition. In television a similar difficulty occurs and there the remedy is to decrease the aperture dimension at right angles to the scanning direction and to increase that parallel to the scanning direction until the ratio is approximately 2 but the area unchanged. The factor of change here involved is

$$\sqrt{2}$$

In the case of the multichannel control signaling system here discussed, we cannot change any aperture width and length but we can do a corresponding thing. Since the interchannel patterns are well transmitted we can use a lesser number of such channels. This makes the channel width greater and corresponds to lengthening the television aperture. By making the channel width greater we will have a lesser number of segments for each rotation of the distributor. Without increasing the speed of the distributor as it passes from segment to segment, we can design the distributor to commutate a given channel more frequently. This corresponds to decreasing the width of the aperture in television. The foregoing adjustments are an exact analogue of aperture dimension adjustment in television.

Of course, where the number of channels have already been reduced to the minimum number which will give the desired degree of intelligible reproduction at the receiving end, we cannot increase the channel width by decreasing the number of channels, as above suggested. In this case we will have to leave the number of channels and the channel width the same, and accomplish the result by increasing the speed of the commutator so as to obtain a greater number of commutations per wave length.

In picture transmission considerable study has been given to the distortion due to scanning and the best aperture shape for holding the effect of such distortion to a minimum by proper choice of the shape of the scanning aperture. This is thoroughly discussed in the article of Pierre Mertz and Frank Gray in the Bell System Technical Journal of July 1934, pages 465–515, entitled "A theory of scanning and its relation to the transmitted signal in telephotography and television". These distortions are of two types, blurring and bringing in extraneous patterns in pictures. They are also present in this proposed system in their corresponding forms. Blurring becomes the transfer of energy from one band to another and the extraneous picture patterns become extraneous speech patterns. While the optimum arrangement in picture transmission was obtained by aperture adjustment and there is no aperture here to adjust, yet methods are available for making the corresponding adjustments where the cost of doing so is small enough and the advantage great enough to warrant doing it. These methods will now be discussed.

In television scanning theory it is known that it is desirable for minimum distortion to scan with sine beams having a cross-section such as shown at the left of Fig. 17, either at the top or the bottom of the figure. At the top is represented a cross-section of a double sine beam and at the bottom a cross-section of a single sine beam is portrayed. By the use of a beam of this character we get, in addition to the main line of scanning shown from a to b, a slight pick-up from the previous line due to a'—a and also from the succeeding line due to b'—b. In picture transmission these edges give the effect of weighting the preceding and succeeding sequence at this point. A similar result can be obtained by the use of a discharge circuit as shown in Fig. 18. Here, a series resistance R_s is included in series with the main line and across the main line a condenser C_s is connected in parallel with a shunt resistance R_s'. By a proper design of these elements the discharge rate of the condenser C in the integrator circuit can be controlled. This enables the condenser C to discharge at such a rate as to retain a portion of its charge from the previous time interval. By choosing proper values of R_s, R_s' and C, as much of this effect can be obtained as desired. Of course, the ground brush for discharging the condenser at the end of the integrating period is omitted in this case.

As already stated, by the use of the sine shaped scanning beam in picture work, pick-up is obtained from the lines in both sides of the main line being scanned, as represented in the diagram of Fig. 17 at the left. Thus, we get the main area in line M but fringes in P and S. The circuit of Fig. 18 produces a result equivalent to this, as shown in Fig. 17 at X. Here the rectangular appearance of the figure does not have the same significance as in picture transmission where both dimensions are length. It is used here merely for convenience and the dimensions depend on the scale chosen for amplitude and frequency band, the vertical dimension representing amplitude and the horizontal dimension frequency band.

It should be noted that this figure differs from the diagram just to the left in two respects. It is incremental instead of smooth and is not symmetrical.

The lack of symmetry can be overcome by the use of a delay circuit, in addition to a network such as shown in Fig. 18, for controlling the rate of discharge of the capacity C of the integrator. Such an arrangement is shown in Fig. 19. Here we insert between the pad P_0 and the integrator I_0 two branches. The main branch includes a delay circuit D and the auxiliary branch includes a loss pad LP. The resistance capacity combination in the main line serves to control the discharge of the condenser C as before, so that in addition to pick-up from the line being scanned, some remnant energy from the preceding scanning line will be picked up as shown at P in the Y portion of the diagram of Fig. 17. The delay circuit D in Fig. 19 is so designed as to delay the transmission of the wave form by a time equal to the time elapsing between successive commutations. The result is, that when the wave in the vertical line which is to be scanned arrives through the delay circuit at the integrator, a corresponding portion of the wave in the succeeding line of scanning or commutation is transmitted through the loss pad LP and arrives at the integrator at the same time. The amplitude of the wave from the succeeding line of scanning can be controlled by the pad LP to produce any amount desired. This gives the figure shown at Y in Fig. 17 the projection to the right indicated at S. Thus, it will be seen that the figure representing the complex resultant pick-up is symmetrical.

The same result may be obtained without the use of a network in the main line by the arrangement of Fig. 20. Here three branches are included between the pad P_0 and the integrator I_0. The main branch includes a delay network D equal to the time of one rotation of the commutator. The lower branch includes a loss pad LP just as in the case of Fig. 19. The upper branch includes a delay circuit 2—D and a loss pad LP. The delay circuit 2—D introduces twice the delay of the circuit D. The result is, that along with the energy from the line being scanned, a controlled amount of energy from the succeeding scanning line will be applied to the integrating circuit by the lower branch, while a corresponding amount of energy from the preceding line of scanning will be applied at the same time by the upper branch. In both Figs. 19 and 20, amplifiers may be inserted in the branches to insulate the branches from each other.

The second objection, the lack of smoothness can be overcome to any extent desired by splitting the paths as shown in the lower part of Fig. 20. Thus in the middle branch delay D could be split into three equal parts each having delay D/3, with a pair of wires led off after the first third for delay D/3, another pair after the second third for delay 2D/3 and a third after the whole delay of 3D/3. Then the upper branch delay can be divided as shown to give delay branches of 4D/3, 5D/3 and 6D/3. This would then make a 7-step scanning device. Dividing the delay into D/4 fractions would give nine steps as shown in Fig. 17 at Z. This process can, of course, be carried as far as desired.

As stated before the scanning in the direction of frequency, that is vertically in Fig. 4 leads approximatel y to non-penalized scanning because adjacent time intervals correspond to adjacent time intervals of scanning. To the extent that a frequency band is integrated, however, there is a discontinuity and so penalized scanning. If it is desired to correct this by the equivalence of adjusting the aperture shape, it is obvious that this can readily be done by circuits of the sort described here for picking up earlier and later energy from the same channel. In particular, simple tie-ins of the proper loss can be made from each channel to the adjacent channels or, better yet, to the adjacent' channels excluding the pitch or FP channel. Similarly the RsCsRs' circuit of Fig. 18 will give a tie-in to the preceding channel traversed unless a discharging arrangement is provided before it is connected to the next channel.

In conclusion a word remains to be said in connection with the main line (including the distributors) and the filters in the receiving channels. Referring to Fig. 2, it has already been pointed out that in connection with the pick-up circuit an auxiliary brush G may be provided, which follows the brush S for picking up the signals and serves to momentarily ground the condenser C of the integrating circuit to wipe off any charge that may have remained after the pick-up operation, thereby enabling the condenser to be charged up in accordance with the portion of the wave corresponding to the next integrating period. The brush G cannot, of course, be used where a circuit of the type shown in Fig. 18 is employed as a portion of the remanent charge after the pick-up is allowed to remain for the succeeding pick-up as already described. The auxiliary brush could be used, however, in connection with circuits such as those shown in Figs. 19 and 20.

In the main line amplifiers such as TA and RA are provided. These amplifiers must be direct current amplifiers, as the energy transmitted is of a direct current nature for continuous talking. The sounds themselves have low syllabic frequencies probably as low as one or two cycles per second. In addition to this, the direct current gives the bias to bring the energy to zero at the proper time. For example, referring to Fig. 7, here the upper curve shows the nature of the current transmitted by the amplifier on the assumption that the transmitted wave is a sine wave. With the direct current bias present the current transmitted will be zero at points e and p. If the direct current bias is removed, due to the fact that the amplifier does not transmit direct current, the condition will be as represented by the lower curve. Here it will be seen we have as much current at the former silent intervals as we have at the full talking period n while at o and m we would have no current, whereas, as shown by the upper curve, there should be current transmitted at these points. Obviously then, it is necessary to transmit the direct current component. The gain of the amplifier should be sufficient to make up for the loss due to the integrating circuits. If the line cannot handle direct current a carrier frequency shift may be used or means may be provided for resupplying it at the receiving end.

Coming now to the receiving end of the scanning circuit, the problem is two-fold. First, the proper signal impulse must be distributed to the proper channel and, second, the pulse effect of commutation should be smoothed out. The first problem involves synchronism which is simply arranged by adjusting the two rings of the distributor segments relative to one another. This may be done by transmitting synchronizing currents in accordance with well known methods.

In order to smooth out the pulse effect, low-pass filters F_{30}' to F_{40}' are provided as shown in Fig. 3. The filters serve to store up the momentary potential applied during commutation so that the effect is held over until the next commutation interval.

The charge applied to the commutator segment should be removed before a new charge is applied at the next commutation interval. This may be accomplished by the circuit design, particularly with reference to impedances. The left-hand condenser in the filter circuit F_{30}', for example, will have a low impedance as compared with that of the remainder of the filter network to the right, but by making the impedance looking into the output of the amplifier in the main line a relatively low resistance, the effect of the preceding charge may be overcome. As the brush contacts the commutator segment to apply a new charging potential, if the new charging potential is lower than that remaining upon the condenser from the preceding commutation, the latter will discharge into the low impedance of the amplifier and bring the charge of the condenser down to the level of the new charging potential. If the new charging potential is still higher, however, only the additional potential will be added to the condenser. For example, if the condenser is already charged to two volts and at the next commutation the charging potential is only one volt, the condenser will discharge down to one volt. If the new charging potential is three volts, on the other hand, the charge in the condenser will be built up from two volts to three volts.

It will be obvious that the general principles herein disclosed may be embodied in many other organizations widely different from those illustrated, without departing from the spirit of the invention as defined by the following claims.

What is claimed is:

1. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately

the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, transmitting timed parts of each of said waves in successive order over a transmission medium to the exclusion of waves representing said invariable information, producing artificially waves representing said invariable information, and combining effects of said artifically produced waves and said transmitted waves to reproduce said signal.

2. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, dividing corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, transmitting over a transmission medium in successive order portions of each of said waves corresponding to the same time interval, successively transmitting in the same order the portions of said waves corresponding to the ensuing time intervals, excluding from the transmission medium waves representing said invariable information, producing artifically waves representing said invariable information, and combining effects of said artifically produced waves and said transmitted waves to reproduce said signal.

3. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, dividing corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, transmitting over a transmission medium in successive order portions of each of said waves corresponding to the same time interval, successively transmitting in the same order the portions of said waves corresponding to the ensuing time intervals, excluding from the transmission medium waves representing said invariable information, separating wave portions of different transmitted waves from each other and arranging the portions of each wave in proper timed relation so as to produce a wave corresponding to each original wave, producing artificially waves representing said invariable information, and combining effects of said artificially produced waves and said transmitted waves to reproduce said signal.

4. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, dividing corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, integrating the portions of each wave corresponding to each timed interval, transmitting over a transmission medium in successive order integrated values of portions of each of said waves corresponding to the same timed interval, successively transmitting in the same order the portions of said waves corresponding to the ensuing time intervals, excluding from the transmission waves representing said invariable information, producing artificially waves representing said invariable information, and combining effects of said artificially produced waves and said transmitted wave values to reproduce said signal.

5. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, dividing corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, integrating the portions of each wave corresponding to each timed interval, transmitting over a transmission medium in successive order integrated values of portions of each of said waves corresponding to the same timed interval, successively transmitting in the same order the portions of said waves corresponding to the ensuing time intervals, excluding from the transmission waves representing said invariable information, separating the transmitted integrated values of portions of different waves from each other and arranging the separated integrated values for each wave in proper timed relation so as to produce a wave corresponding to each original wave, producing artificially waves representing said invariable information, and combining effects of said artificially produced waves and said transmitted wave values to reproduce said signal.

6. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, dividing corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, integrating the portions of each wave corresponding to each timed interval, transmitting over a transmission medium in successive order integrated values of portions of each of said waves corresponding to the same timed interval, successively transmitting in the same order the portions of said waves corresponding to the ensuing time intervals, excluding from the transmission waves representing said invariable information, selecting the transmitted integrated values of portions of different waves from each other into separate channels, arranging the integration values of portions of the same wave in proper timed relation in each channel, holding over each integrated value during the time interval in which it was integrated so as to produce an uninterrupted wave corresponding to each original wave, producing artificially waves representing said invariable information, and combining effects of said artificially produced waves and said transmitted wave values to reproduce said signal.

7. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, selecting said defining waves into separate channels, scanning the values of the waves in the several channels in regular order and then successively scanning later values of the same waves from channel to channel until the waves in the channels have been completely scanned, transmitting the scanned values over a common transmission medium, excluding from the transmission medium waves representing said invariable information, producing artificially waves representing said invariable information, and combining effects of said artificially produced waves and said scanned wave values to reproduce said signal.

8. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, selecting said defining waves into separate channels, scanning the values of the waves in the several channels in regular order and then successively scanning later values of the same waves from channel to channel until the waves in the channels have been completely scanned, transmitting the scanned values over a common transmission medium, excluding from the transmission medium waves representing said invariable information, selecting the transmitted scanned elements of the different waves from each other into separate channels, the scanned elements of each wave being selected into the corresponding channel in the order in which they were scanned along the wave so as to reproduce a wave corresponding to the original wave, producing artificially waves representing the invariable information of the original signal, and combining the effects of said artificially produced waves and said reproduced waves to reproduce said signal.

9. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, the frequency of said defining waves being below audibility, dividing corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, transmitting over a transmission medium in successive order portions of each of said waves corresponding to the same time interval, successively transmitting in the same order the portions of said waves corresponding to the ensuing time intervals, excluding from the transmission medium waves representing said invariable information, whereby the frequency range employed for transmission over said medium is only a fractional part of the band width required for the original signal, producing artificial waves representing said variable information, and combining effects of said artificially produced waves and said transmitted waves to reproduce said signal.

10. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, the frequency of said defining waves being below audibility, selecting said defining waves into separate channels, scanning the values of the waves in the several channels in regular order and then successively scanning later values of the same waves from channel to channel until the waves in the channels have been completely scanned, transmitting the scanned values over a common transmission medium, excluding from the transmission medium waves representing said invariable information, whereby the frequency range employed for the transmission over said medium is only a fractional part of the band width required for the original signal, producing artificially waves representing said invariable information, and combining effects of said artificially produced waves and said scanned wave values to reproduce said signal.

11. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to transmit timed parts of each of said waves in successive order over a transmission medium to the exclusion of waves representing said invariable information, means for producing artificially waves representing said variable information, and means to combine effects of said artificially produced waves and said transmitted waves to reproduce said signal.

12. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to divide corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, means to transmit over a transmission medium in successive order portions of each of said waves corresponding to the same time interval, means to successively transmit in the same order the portions of said waves corresponding to the ensuing time intervals, means to exclude from the transmission waves representing said invariable information, means to produce artificially waves representing said invariable information, and means to combine effects of said artificially produced waves and said transmitted waves to reproduce said signal.

13. In a signaling system, means to produce a signal containing variable information and in-

variable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to divide corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, means to transmit over a transmission medium in successive order portions of each of said waves corresponding to the same time interval, means to successively transmit in the same order the portions of said waves corresponding to the ensuing time intervals, means to exclude from the transmission waves representing said invariable information, means to separate wave portions of different transmitted waves from each other and to arrange the portions of each wave in proper timed relation so as to produce a wave corresponding to each original wave, means to produce artificially waves representing said invariable information, and means to combine effects of said artificially produced waves and said transmitted waves to reproduce said signal.

14. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to divide corresponding portions of the waves representing a syllable into a plurality of timed intervals, means to integrate the portion of each wave corresponding to each timed interval, means to transmit over a transmission medium in successive order integrated values of portions of each of said waves corresponding to the same time interval, means to successively transmit in the same order the portions of said waves corresponding to the ensuing time intervals, means to exclude from the transmission waves representing said invariable information, means to produce artificially waves representing said invariable information, and means to combine effects of said artificially produced waves and said transmitted wave values to reproduce said signal.

15. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to divide corresponding portions of the waves representing a syllable into a plurality of timed intervals, means to integrate the portion of each wave corresponding to each timed interval, means to transmit over a transmission medium in successive order integrated values of portions of each of said waves corresponding to the same time interval, means to successively transmit in the same order the portions of said waves corresponding to the ensuing time intervals, means to exclude from the transmission waves representing said invariable information, means to separate the transmitted integrated values of portions of different waves from each other and to arrange the separated integrated values for each wave in proper timed relation so as to produce a wave corresponding to each original wave, means for producing artificially waves representing said invariable information, and means for combining effects of said artificially produced waves and said transmitted wave values to reproduce said signal.

16. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to divide corresponding portions of the waves representing a syllable into a plurality of timed intervals, means to integrate the portion of each wave corresponding to each timed interval, means to transmit over a transmission medium in successive order integrated values of portions of each of said waves corresponding to the same time interval, means to successively transmit in the same order the portions of said waves corresponding to the ensuing time intervals, means to exclude from the transmission waves representing said invariable information, means to select the transmitted integrated values of portions of different waves from each other into separate channels, means to arrange integrated values of portions of the same wave in proper timed relation in each channel, means for holding over each integrated value during the time interval in which it was integrated so as to produce an uninterrupted wave corresponding to each original wave, means for producing artificially waves representing said invariable information, and means to combine effects of said artificially produced waves and said transmitted wave values to reproduce said signal.

17. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to select said defining waves into separate channels, a distributor for scanning the values of the waves in the several channels in regular order and then successively scanning later values of the same waves from channel to channel until the waves in the channels have been completely scanned, means for transmitting the scanned values over a common transmission medium, means for excluding from the transmission medium waves representing said invariable information, means for producing artificially waves representing said invariable information, and means for combining effects of said artificially produced waves and said scanned wave values to reproduce said signal.

18. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom

of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to select said defining waves into separate channels, a distributor for scanning the values of the waves in the several channels in regular order and then successively scanning later values of the same waves from channel to channel until the waves in the channels have been completely scanned, means for transmitting the scanned values over a common transmission medium, means for excluding from the transmission medium waves representing said invariable information, a distributor for selecting the transmitted scanned elements of the different waves from each other into separate channels, the scanned elements of each wave being selected into the corresponding channel in the order in which they were scanned along the wave so as to reproduce a wave corresponding to the original wave, means for producing artificially waves representing the invariable information of the original signal, and means for combining the effects of said artificially produced waves and said reproduced waves to reproduce said signal.

19. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, the frequency of said defining waves being below audibility, means for dividing corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, means for transmitting over a transmission medium in successive order portions of each of said waves corresponding to the same time interval, means for successively transmitting in the same order the portion of said wave corresponding to the ensuing time interval, means for excluding from the transmission medium waves representing said invariable information, whereby the frequency range employed for transmission over said medium is only a fractional part of the band width required for the original signal, means for producing artificially waves representing said invariable information, and means for combining effects of said artificially produced waves and said transmitted waves to reproduce said signal.

20. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, the frequency of said defining waves being below audibility, means for selecting said defining waves into separate channels, a distributor for scanning the values of the waves in the several channels in regular order and then successively scanning later values of the same waves from channel to channel until the waves in the channels have been completely scanned, means for transmitting the scanned values over a common transmission medium, means for excluding from the transmission medium waves representing said invariable information, whereby the frequency range employed for transmission over said medium is only a fractional part of the band width required for the original signal, means for producing artificially waves representing said invariable information, and means for combining effects of said artificially produced waves and said scanned wave values to reproduce said signal.

21. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to divide corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, an integrator for integrating the portion of each wave corresponding to each timed interval, and means for applying the integrated portion of each wave corresponding to a given time interval to a transmission medium together with the portion of each wave corresponding to an adjacent time interval.

22. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to divide corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, an integrator for integrating the portion of each wave corresponding to each timed interval, and means for applying the integrated portion of each wave corresponding to a given time interval to a transmission medium together with the portion of each wave corresponding to a preceding timed interval.

23. In a signaling system, means to produce a signal containing variable information and invariable information represented by a complex wave, means to analyze said wave and derive therefrom a simple set of parameters having approximately the number of degrees of freedom of the variable elements of said signal source, means to translate said set of parameters into a set of defining waves that respectively define the variations of said parameters, means to divide corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, an integrator for integrating the portion of each wave corresponding to each timed interval, and means for applying the integrated portion of each wave corresponding to a given time interval to a transmission medium together with portions of the wave corresponding to the preceding and the succeeding time intervals.

24. In a privacy system, the method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the

variations of said parameters but are not in themselves intelligible, transmitting timed parts of each of said unintelligible waves in successive order over a transmission medium to the exclusion of waves representing said invariable information, producing artificially waves representing said invariable information and combining effects of said artificially produced waves and said transmitted unintelligible waves to reproduce said signal.

25. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, the frequency of said defining waves being below audibility, dividing corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, transmitting over a transmission medium in successive order portions of each of said waves corresponding to the same time interval, successively transmitting in the same order the portions of said waves corresponding to the ensuing time interval, excluding from the transmission medium waves representing said invariable information so that the frequency range employed for transmission over said medium is only a fractional part of the band width required for the original signal, transmitting the band of fractional width at energy levels of the order of the energy levels which would exist if the full signal band width were transmitted, so that a higher ratio of signal to noise results within the band transmitted, producing artificial waves representing said invariable information, and combining effects of said artificially transmitted waves and said transmitted waves to reproduce said signal.

26. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in deriving from the complex wave a simple set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, the frequency of said defining waves being below audibility, dividing corresponding portions of each of the waves representing a syllable into a plurality of timed intervals, transmitting over a radio transmission medium in successive order portions of each of said waves corresponding to the same time interval, successively transmitting in the same order the portions of said waves corresponding to the ensuing time intervals, excluding from the radio transmission medium waves representing said invariable information, so that the frequency range employed for transmission over said radio transmission medium is only a fractional part of the band width required for the original signal, thereby reducing the effect of fading in the band of fractional width transmitted, producing artificial waves representing said variable information, and combining effects of said artificially produced waves and said transmitted waves to reproduce said signal.

27. The method of reproducing a signal containing variable information and invariable information and represented by a complex wave, which consists in instantaneously analyzing the complex wave into a set of parameters having approximately the number of degrees of freedom of the variable elements of the signal source, deriving from said set of parameters a set of defining waves that respectively define the variations of said parameters, transmitting timed parts of each of said waves in successive order over a transmission medium to the exclusion of waves representing said invariable information, producing artificially waves representing said invariable information, and combining effects of said artificially produced waves and said transmitted waves to reproduce said signal.

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