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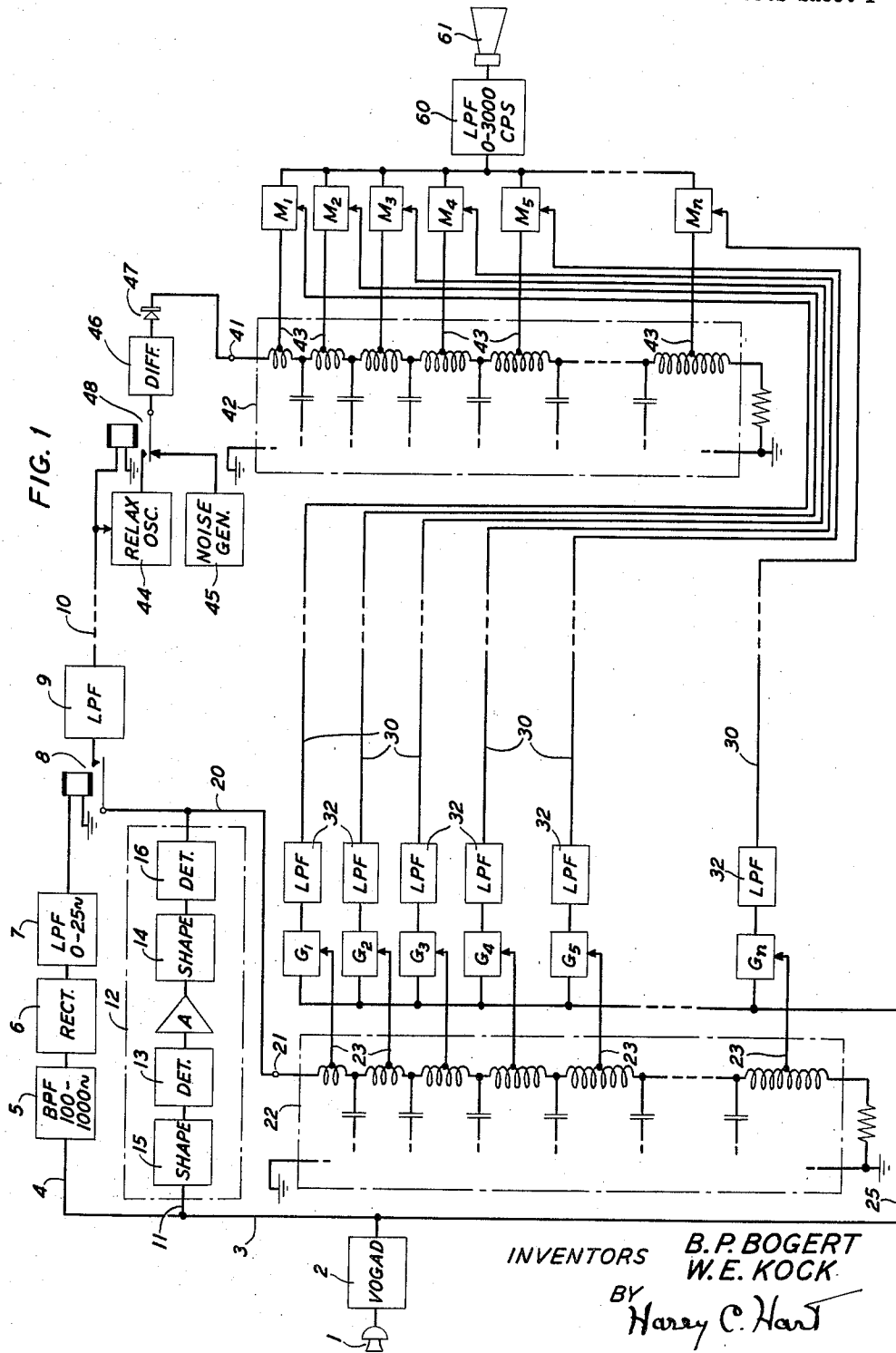
B. P. BOGERT ET AL

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NARROW BAND TRANSMISSION OF SPEECH

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2 Sheets-Sheet 1



INVENTORS
BY *Harry C. Hart*
ATTORNEY

B. P. BOGERT
W. E. KOCK

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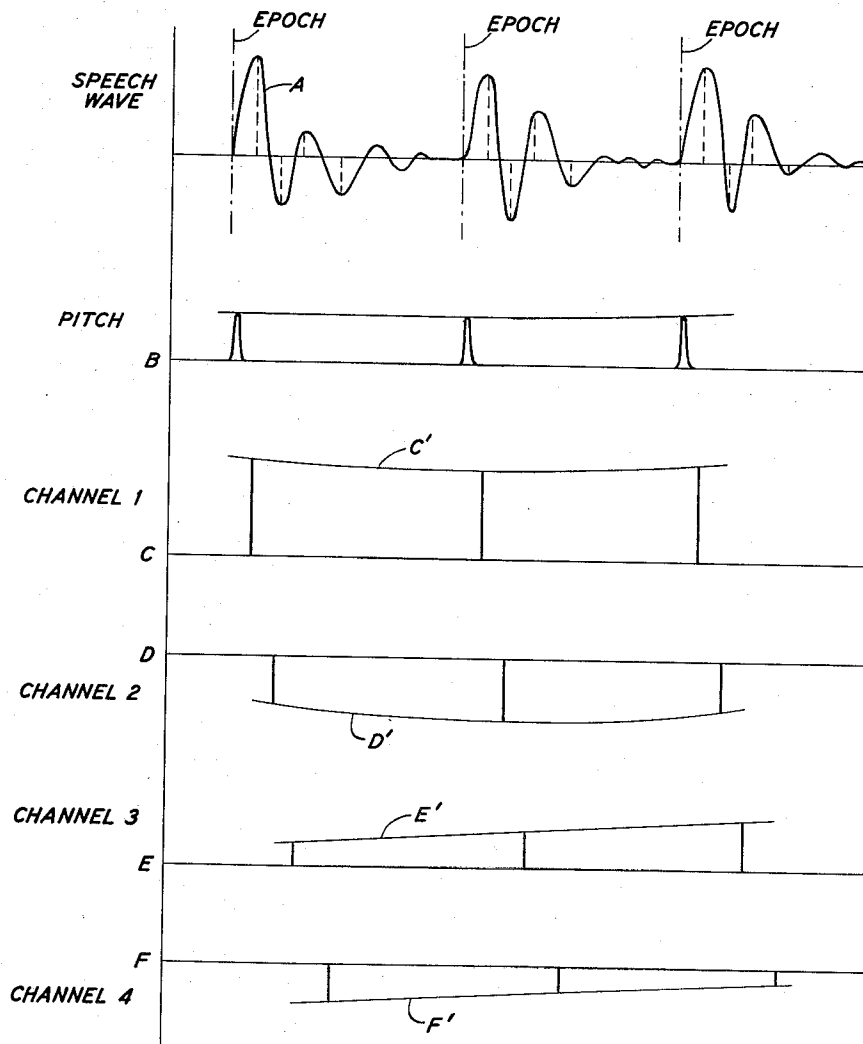
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FIG. 2



INVENTORS B. P. BOGERT
W. E. KOCK

BY *Harry C. Hart*
ATTORNEY

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NARROW BAND TRANSMISSION OF SPEECH

Bruce P. Bogert, Morristown, and Winston E. Kock, Basking Ridge, N.J., assignors to Bell Telephone Laboratories, Incorporated, New York, N.Y., a corporation of New York

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This invention relates to frequency band compression and its principal object is to compress the band of frequencies required for telephony.

It is recognized that many of the amplitude variations of a speech wave, which in the technical sense carry "information," are redundant in that they are of little use to the listener in determining the meaning of the speaker. This knowledge has been turned to account in the construction of frequency band compression systems of several kinds, notably those of Dudley Patent 2,115,803, May 3, 1938, and Dudley Patent 2,151,091, March 21, 1939. The system of the first of these two patents is based upon the recognition that a speech wave changes its character but little from each fundamental period to the next; i.e., it is almost a periodic function. Dudley turns this situation to account by discarding some periods of the wave, transmitting the others on a scale which is expanded in time and compressed in frequency and, at the receiver station, restoring each transmitted wave to its original dimensions and filling in the gaps with repetitions of the transmitted wave. In the system of the second Dudley patent referred to above, the entire spectrum of an original speech wave is broken down into a number of subbands by the use of narrow band-pass filters. A control signal representing the energy in each of these subbands is transmitted to a receiver station where it operates, in conjunction with local sources of periodic and aperiodic waves, to control the production of synthetic speech.

It is evident that the former system is open to the objection that, for each period of the wave which is omitted, the abruptness of the change from the last repetition of its predecessor to the first reproduction of its successor is doubled. The objections to the second system are rather of a technological nature, i.e., each of the narrow band-pass filters required for frequency analysis of the voice poses a series design problem, and the problem is only increased as the number of filters required is increased.

It is well known in the field of multiplex telephony that independent messages may be transmitted in two alternative ways, i.e., by frequency or wire separation or on a time division basis. In a time division multiplex system the voice waves on a number of independent channels are sampled in succession at a rate sufficiently high to carry the information contained in each message. This rate, known as the "Nyquist rate" is twice the highest component frequency to be transmitted.

The sequence of amplitudes of a number of independent voice waves taken by a time division multiplex sampler in the course of one of its rotation cycles is an arbitrary wave. Such a system makes no use of the important fact that substantial correlation exists between each full period of a speech wave and the next. Hence, while time division multiplex telephony offers many advantages, frequency band compression is not among them.

The present invention turns the near-periodic character

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of a speech wave to account in the compression of the frequency band required to transmit it. It does this in the following fashion:

The fundamental frequency and/or period of the speech wave to be transmitted is first determined, and a marker pulse or other electrical indication is produced at the inception of each fundamental period. This marker pulse in turn generates a train of a preassigned number of sampling pulses, and this train terminates before the termination of the voice wave period in which it starts. Each pulse of this train takes a sample of the amplitude of the speech wave, and these several samples are individually transmitted over independent channels to a receiver station. These events are repeated for the next and every other following full fundamental period of the speech wave. Thus, channel No. 1 carries the No. 1 samples of all the periods but no No. 2 samples of any period; channel No. 2 carries all the No. 2 samples of all the periods but no No. 1 samples of any period, and so on.

Now, because of the almost periodic character of the speech wave, the No. 1 samples are nearly alike from period to period; the No. 2 samples are likewise nearly alike from period to period although they may differ markedly from the No. 1 samples; the No. 3 samples are nearly alike from period to period although they may differ markedly from No. 1 samples and from No. 2 samples, and so on. Thus, each channel carries samples which are nearly alike in magnitude and change only very slowly over a considerable number of periods. Hence, each of these channels may have a very narrow pass-band, e.g., 15 cycles per second or less, and the control signals which it carries are of correspondingly low frequencies.

At the receiver station a relaxation oscillator or other pulse source initiates a train of pulses of number and spacing like the number and spacing of the train generated at the transmitter station. The rate at which successive ones of these pulse trains is generated is controlled as by a transmitted pitch control signal, in a fashion to remain at all times substantially equal to the fundamental pitch of the speech wave. The transmitted low frequency control signals actuate modulators which admit pulses of this locally generated train in sequence to a common reproducer, with amplitudes proportional to the several control signals. As a result the reproducer receives, for each full fundamental period of the speech wave, a train of pulses whose instants of occurrence and whose amplitudes are substantially like those of the speech samples originally taken at the transmitter station. The averaging process necessarily introduced by any physically realizable apparatus such as the reproducer then converts this pulse sequence into a replica of the original speech wave.

Because of the nature of a speech wave, namely, that each period is started by a puff of air from the vocal cords which is reflected in a major peak at the beginning of the period, it is advantageous to bunch the samples more densely at the beginning of each period and less densely at its end. The invention provides convenient means for securing this nonuniform distribution in time of the sampling pulses of each train while retaining equality in all respects of each train with every other train.

The invention will be fully apprehended from the following detailed description of a preferred embodiment hereof taken in connection with the appended drawings in which:

Fig. 1 is a block schematic diagram illustrating a narrow band transmission system in accordance with the invention; and

Fig. 2 shows a group of waveform diagrams of assistance in the exposition of the invention.

Referring now to Fig. 1, a speech signal which may be derived through a vogad 2 from a source such as a microphone 1 is passed by way of a conductor 3 into three parallel branches. The upper branch 4, which comprises a band-pass filter 5, a rectifier 6, a low-pass filter 7 and the winding of a relay 8 connected in tandem, serves as a voiced sound recognizer. As is well known, if the bandwidth of the filter 5 be selected to embrace the principal components of a voiced sound, e.g., if the filter be proportioned to pass frequencies in the range 100 c.p.s.—1000 c.p.s., and if the low-pass filter 7 be adjusted to pass only syllabic frequencies, the relay 8 is operated by voiced sounds and remains unoperated when the sounds picked up by the microphone 1 are unvoiced. Closure of the contacts of the relay 8 by voiced sounds operates to establish a path for a pitch control signal, derived in the second path 11, through a low pass filter 9, to a transmission medium 10.

The second path 11 comprises a period marker signal generator 12 which may advantageously be of the type which forms the subject matter of E. Peterson Patent 2,593,694 and which is further described by O. O. Gruenz and L. O. Schott in an article published in the Journal of Acoustical Society of America for September 1949 (volume 21), page 487. The principal feature of this generator 12 is a detector 13 followed in tandem by a shaping network 14 which accentuates the amplitude of low frequency components at the expense of higher harmonic component amplitudes. Preferably, each of these steps is carried out two or more times in succession and all of them may be preceded by an auxiliary shaping step as by a network 15. With this arrangement, as is more fully explained in the patent and publication referred to above, the output of the generator 12 comprises a single sharp spike of current which occurs at the instant of the major peak of each wave period, or between the principal zero and the major peak. For present purposes, the principal zero is the last zero value or axis crossing of the speech wave preceding each of its major peaks.

This marker pulse thus indicates the instant of inception of each full period of the speech wave and the frequency at which such pulses are repeated is the fundamental pitch frequency of the speech wave. It is transmitted, as stated above, by way of the low-pass filter 9 to a transmission medium 10 for use at a receiver station as described below.

The same marker pulse is also transmitted by way of a conductor 20 to the input terminal 21 of a wave transmission device 22 such as an electromagnetic or acoustic delay line which is terminated at its far end to prevent reflections. This line 22 is provided with a number of lateral taps 23. It operates to produce a replica of the input marker pulse on these several taps in succession, each at an instant determined by the location of the tap.

The pulses thus appearing on these several taps 23 are applied to the control terminals, indicated by arrow heads, of a number of sampling gates $G_1, G_2 \dots G_n$ to whose conduction terminals the voice signal to be transmitted is applied by way of the third parallel path 25. These sampling gates G_1, G_2 , etc., operate in well known fashion to derive brief samples of the instantaneous amplitudes of the speech wave, each at an instant determined by the location on the delay line of the tap 23 which controls it.

Fig. 2 shows in curve A three successive periods of a typical voice wave. It will be noted that each period differs slightly from its predecessor in amplitude, wave-shape, and duration. As explained above, the apparatus of the invention takes samples of this wave at preassigned instants following its principal zero. A first, a second,

a third and a fourth such sample are shown for each of the three periods.

Curve B shows a sequence of period marker pulses which, as explained in the patent and publication referred to above, are brief sharp spikes, one for each period. Each of these spikes initiates a train of sampling pulses.

Curve C shows the No. 1 samples of all three periods, curve D shows the No. 2 pulses of all three periods, curve E shows the No. 3 pulses of all three periods, and curve F shows the No. 4 pulses of all three periods. As indicated by the envelopes C', D', E' and F' the rate of change of amplitude of the samples of any one of these sets is very small and this means that a correspondingly narrow transmission channel suffices to carry the information which they contain.

Accordingly, the output terminals of the several sampling gates G_1, G_2 , etc., are connected by way of independent channels 30 to a transmission medium, and the narrow bands of the several channels are symbolized by the inclusion, in tandem with each such channel, of a low-pass filter 32. Each of these low-pass filters 32 may be proportioned to pass only syllabic frequencies, i.e., frequencies in the range 0–15 cycles per second.

As shown in the figure, the lateral taps 23 are spaced nonuniformly from end to end of the delay line 22, being somewhat bunched toward the input terminal 21 and somewhat more widely spaced toward the tail end of the line. In consequence the amplitude samples of the speech wave which are taken by the pulse outputs of these several taps 23 in the fashion described above are likewise bunched more closely in the early part of each period and less closely in the later part. While this is in no way essential to the invention it offers the advantage that, for a given number of transmission channels 30 and hence a given number of samples per period of the speech wave, the early part of each period, in which its energy is greatest, is sampled with high resolution and the later part with lower resolution. In other words, the samples are located on the time scale where the need for them is greatest.

Transmission of the resulting control signals to a receiver station may be carried out in any desired fashion, resort being had to modulation or coding techniques if preferred. At the receiver station, after such detection, decoding or other operation as may be required to restore them to their original form, each of these control signals is applied to the control terminal of one of a like number of modulators $M_1, M_2 \dots M_n$. The conduction terminals of these several modulators are connected to the several taps of a delay line 42 which may be identical with the delay line 22 at the transmitter station. This delay line 42 is supplied at its input terminal 41 with a sequence of sharp pulses whose recurrence rate is equal to the fundamental pitch of the voice wave. These pulses may be derived in well known fashion from a pulse source 44 or 45 through a differentiator 46, a rectifier 47, and the contacts of a relay 48. Again, as is well known, the transmitted pitch control signal maintains steady control of the frequency of oscillation of the source 44 in order that it shall continuously be substantially the same as the fundamental frequency of voiced sounds as determined by the period marker signal generator 12. When, as in the case of unvoiced sounds, the pitch signal fails, the coil of the relay 48 is de-energized and the noise source 45 is connected through the back contacts of the relay 48, the differentiator 46 and the rectifier 47 to the delay line 42.

The output terminals of the modulators M_1, M_2 , etc., are connected together and by way of a low-pass filter 60 to a sound reproducer 61. The filter 60 is proportioned to pass frequencies of the entire voice range of interest, e.g., from zero to 3000 c.p.s. As a practical matter, the characteristics of the reproducer 61 may provide all the filtering which may be needed.

With these connections the pulses which appear in succession and with like amplitudes at the several taps 43 of the delay line 42 are individually modulated in amplitude in proportion to the magnitudes of the several control signals. After such modulation the amplitude and spacing on the time scale of each pulse is substantially identical with the amplitude and spacing of the corresponding sample of the original speech wave derived at the transmitter station. Accordingly, this pulse sequence may be reconverted into the original speech wave merely by imposing on it an average process as by the interposition of the filter 60.

The system hereinabove described utilizes a number of independent channels whose individual bandwidths are no greater than the pass bands of the control signal channels of Dudley Patent 2,151,091. Furthermore, for comparable quality of reproduction the number of channels required in the present system is no greater than the number of control signal channels required by the Dudley system. Of principal interest is the fact that the pass band of each channel of the present system extends from zero frequency to a low nonzero frequency, i.e., it may comprise a low-pass filter. This is to be compared with the number of far more intricate and costly band-pass filters required by the system of the Dudley patent.

Various straightforward extensions and modifications of the illustrative embodiment of the invention herein described will suggest themselves to those skilled in the art.

What is claimed is:

1. Signal transmission apparatus which comprises, in combination with a source of a speech sound of which the wave differs only slightly from each of a succession of wave periods to the next wave period, means for generating a marker pulse substantially at the instant of inception of each wave period, a wave propagation device having an input terminal and a plurality of lateral taps, means for applying each said marker pulse to said input terminal, thereby to generate sampling pulses on said taps in succession, means including said sampling pulse generating means for deriving a sequence of samples of the wave of said speech sound in each period, a plurality of narrow band transmission channels equal in number to the samples in each said sequence, means for transmitting similarly numbered samples of all said wave periods as a control signal over one of said channels, like means for transmitting other similarly numbered samples as control signals over said other channels, and, at a receiver station, means for generating pulse trains in synchronism with said sample sequences, means for modulating the several pulses of each said pulse train by said several control signals, and means for sequentially applying said pulses as modulated to a common reproducer.

2. Apparatus as defined in claim 1 wherein the spacing of the taps of said propagation device increases progressively along the length thereof from said input terminal.

3. Apparatus as defined in claim 1 wherein the receiver pulse train generating means comprises a wave propagation device having an input terminal and a plu-

ality of lateral taps, the number and spacing of said taps being like the number and spacing of the taps of the first-named wave propagation device.

4. Signal transmission apparatus which comprises, in combination with a source of a speech sound of which the wave differs only slightly from each of a succession of wave periods to the next wave period, means for generating a marker pulse substantially at the instant of inception of each wave period, means for deriving a train of sampling pulses from each said marker pulse, each of said trains enduring less than a wave period, means including said last-named pulse deriving means for deriving a sequence of samples of the wave of said speech sound in each period, a plurality of narrow band transmission channels equal in number to the samples in each said sequence, means for transmitting a first sample of each wave period as a control signal over one of said channels, like means for transmitting a second and third sample as control signals over individual other channels, and, at a receiver station, means for generating pulse trains in synchronism with said sample sequences, means for modulating the several pulses of each said last-named pulse train by said several control signals, and means for sequentially applying said pulses as modulated to a common reproducer.

5. Apparatus as defined in claim 4 wherein the sampling pulse train deriving means comprises means for bunching said sampling pulses with a greater than average density during the early portion of each wave period.

6. Signal transmission apparatus which comprises, in combination with a source of an original signal of which the wave differs only slightly from each of a succession of wave periods to the next wave period, means for deriving a sequence of samples of the wave of said original signal in each period, a plurality of narrow band transmission channels equal in number to the samples in each said sequence, means for transmitting a first sample of each wave period as a control signal over one of said channels, like means for transmitting a second and third sample as control signals over individual other channels, and, at a receiver station, means for artificially reproducing said original wave under control of said control signals.

7. Apparatus as defined in claim 6 wherein the wave sample deriving means comprises means for bunching said wave samples with a greater than average density during the early portion of each wave period.

References Cited in the file of this patent

UNITED STATES PATENTS

2,098,956	Dudley	Nov. 16, 1937
2,522,539	Riesz	Sept. 19, 1950
2,627,541	Miller	Feb. 3, 1953
2,629,017	Dahlbom et al.	Feb. 17, 1953
2,635,146	Stenburg	Apr. 14, 1953
2,705,742	Miller	Apr. 5, 1955
2,810,787	Di Toro et al.	Oct. 22, 1957

FOREIGN PATENTS

484,337	Canada	June 24, 1952
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