

June 27, 1967

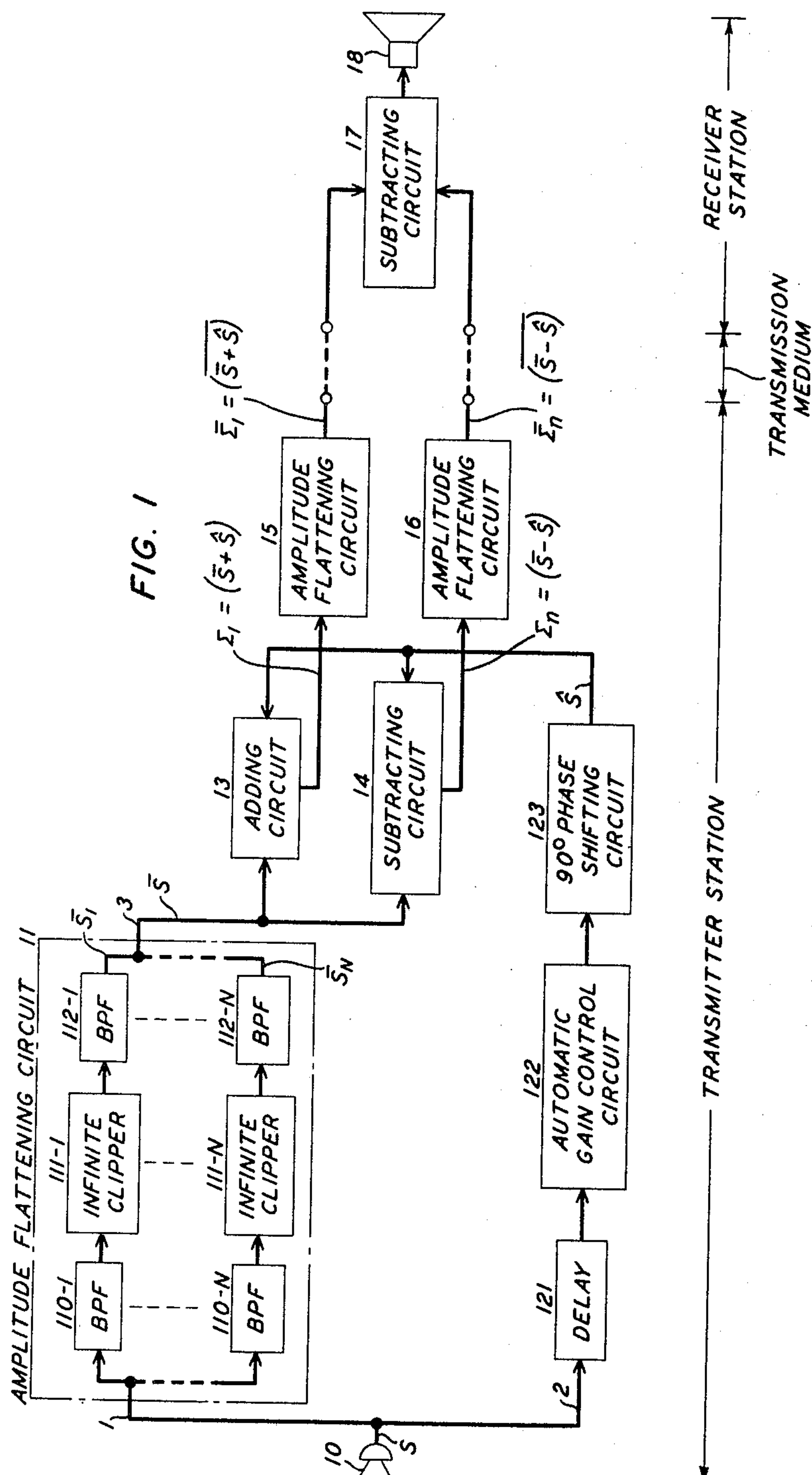
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3,328,526

SPEECH PRIVACY SYSTEM

Filed Dec. 20, 1963

4 Sheets-Sheet 1



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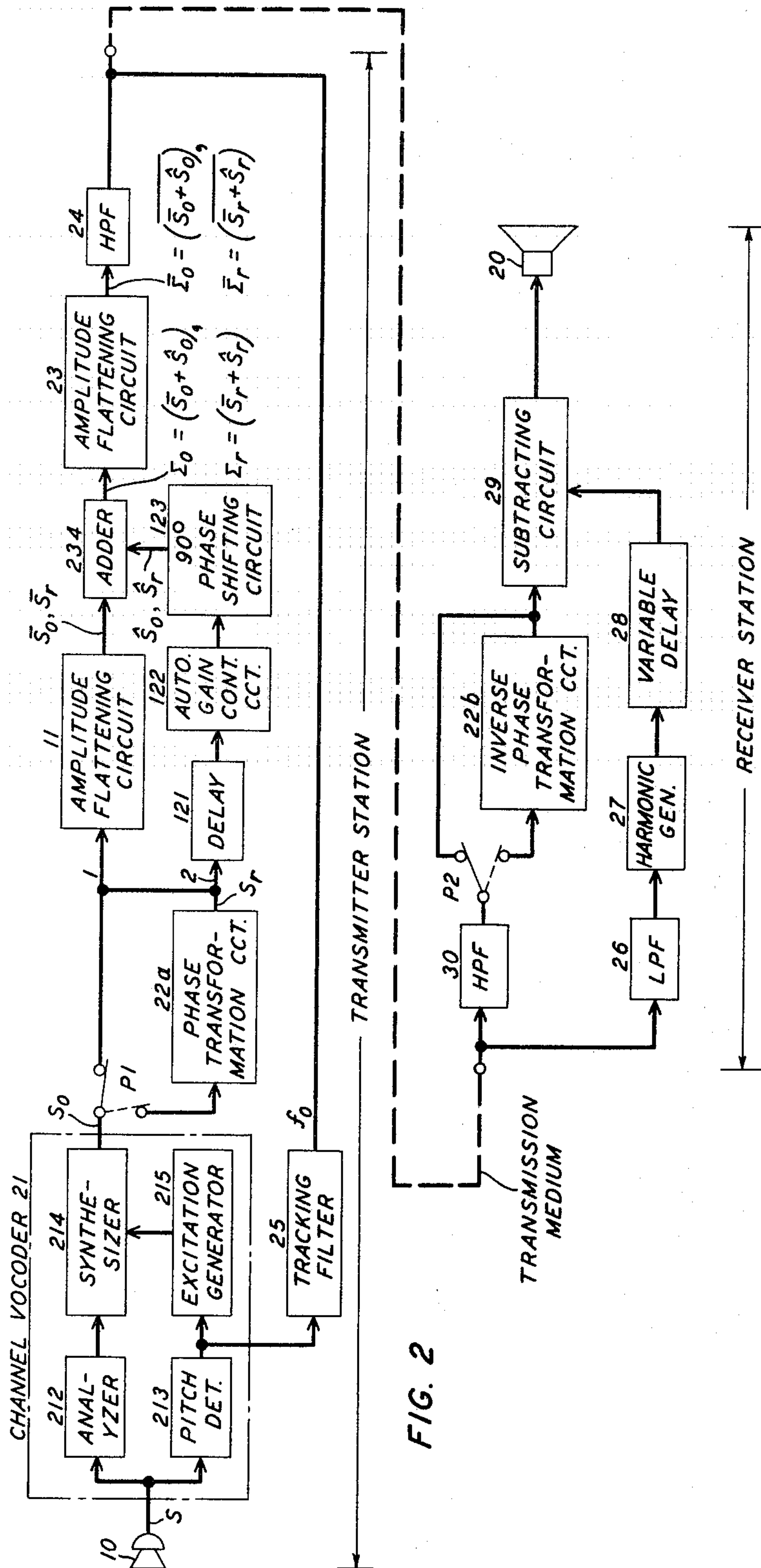
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SPEECH PRIVACY SYSTEM

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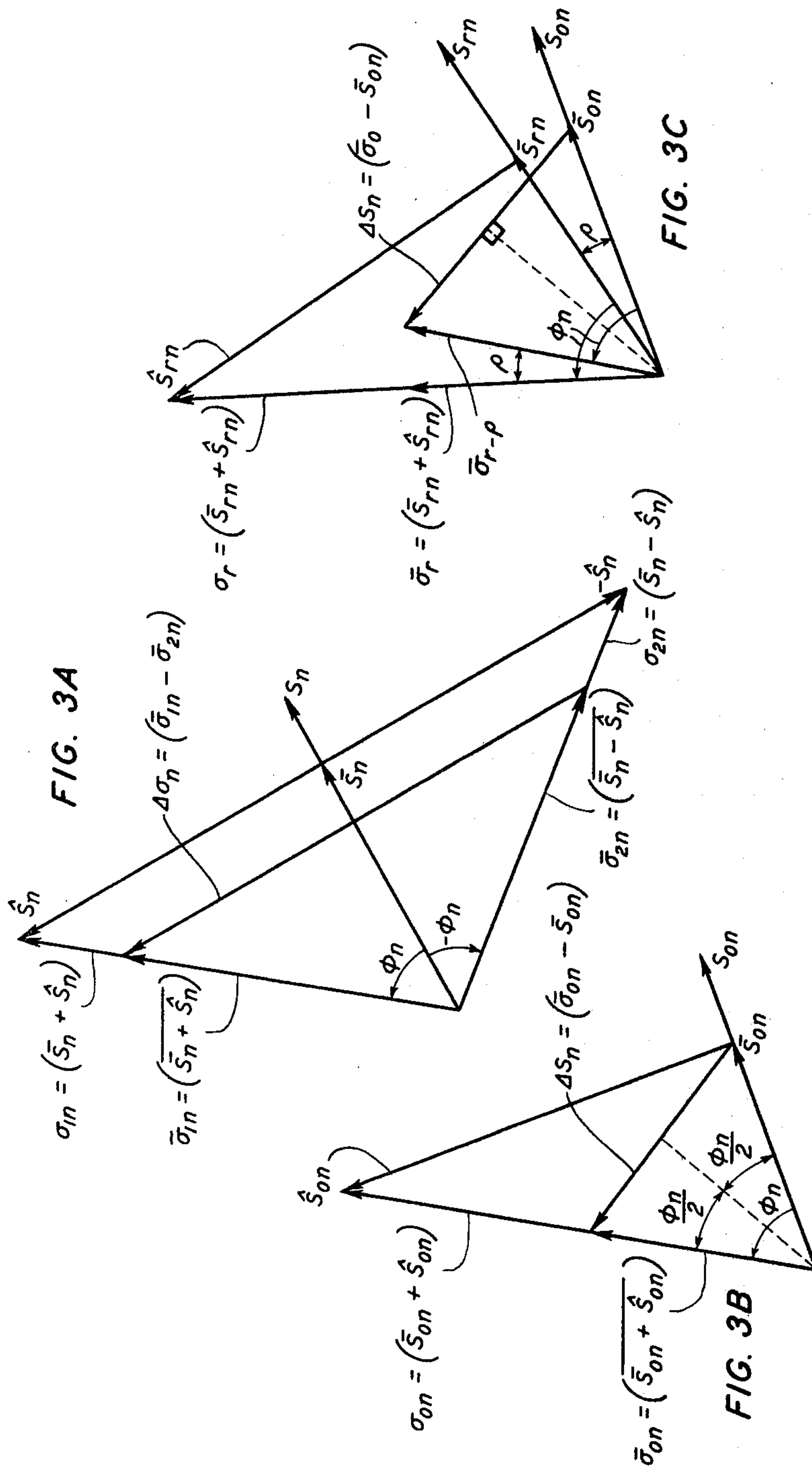
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FIG. 4A

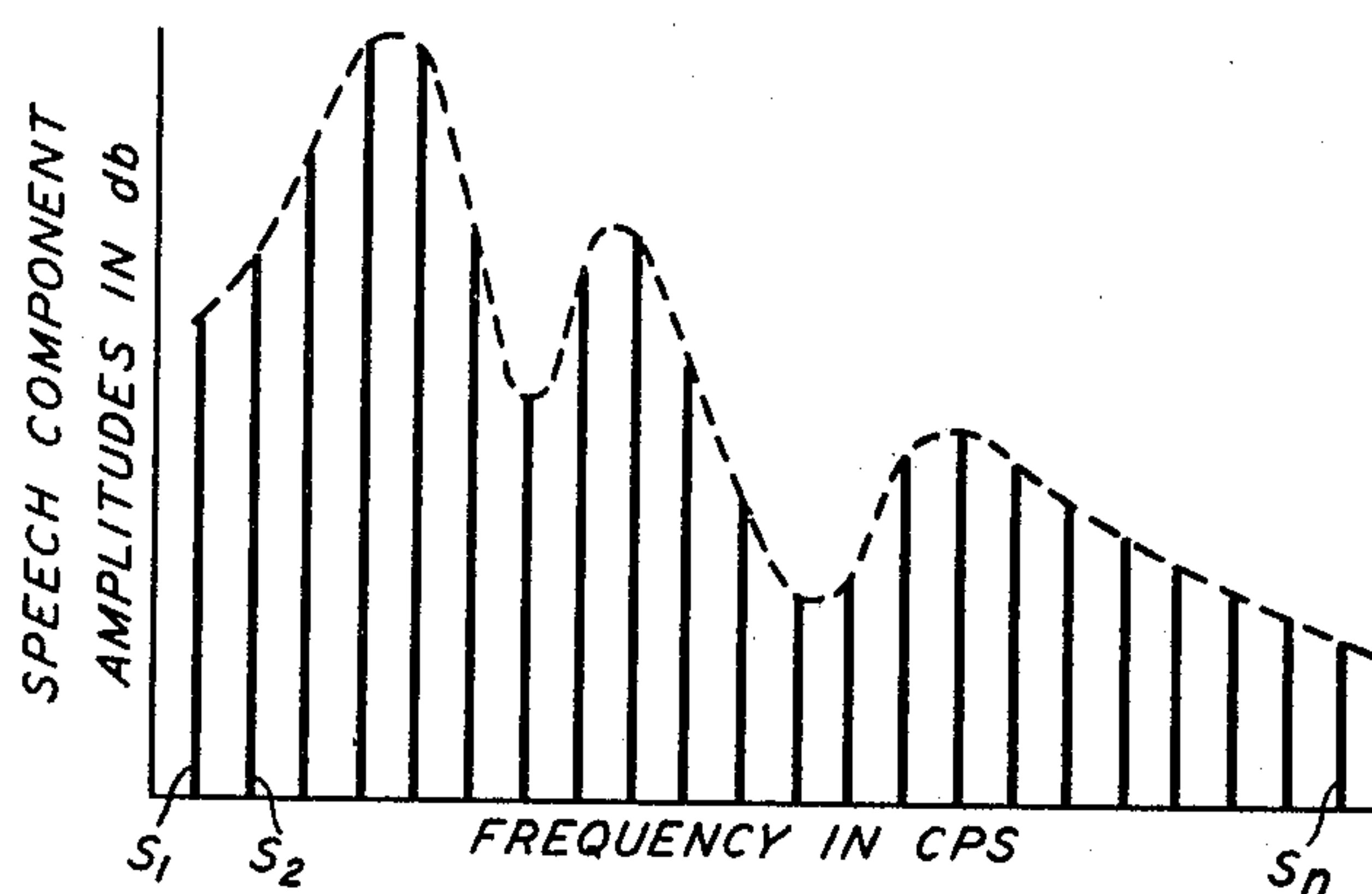
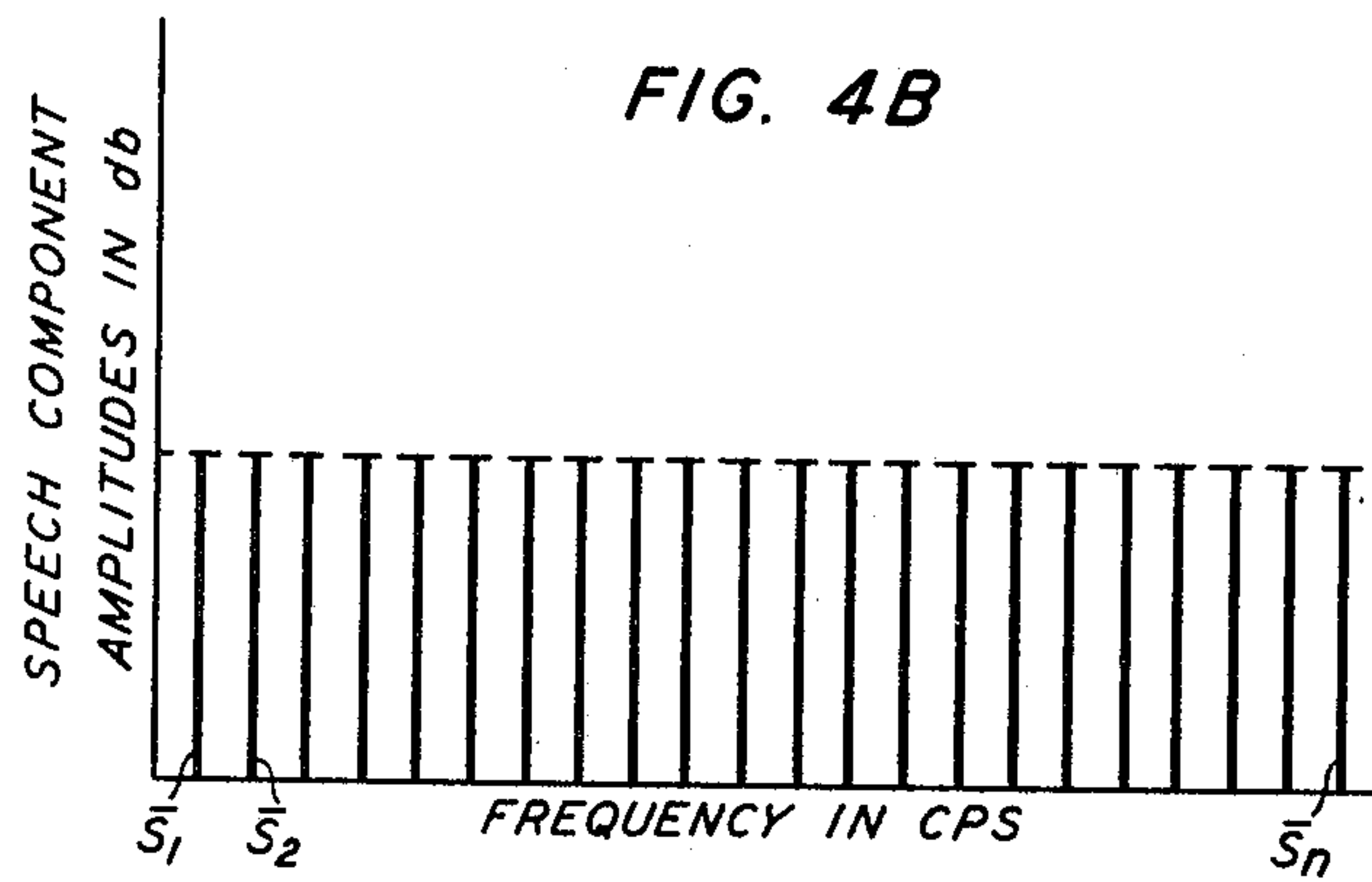


FIG. 4B



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3,328,526

SPEECH PRIVACY SYSTEM

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9 Claims. (Cl. 179—1.5)

This invention relates to the transmission of speech information with privacy.

It is well known that the production of the voiced sounds of human speech may be described in terms of a number of frequency components having various amplitudes and phase angles dependent upon the configuration of the talker's vocal mechanism at a given instant. In the perception of speech, however, it is the amplitudes of speech frequency components which are most important in conveying meaningful information to the human hearing mechanism, while the phase angles of speech frequency components convey relatively little information to the human hearing mechanism. This phenomenon is demonstrated by the well-known vocoder, an example of which is described in H. W. Dudley Patent 2,151,091, issued Mar. 21, 1939, in which only the amplitudes of the frequency components of a speech wave are coded for transmission, and in which the synthesized speech reconstructed from the coded amplitudes has frequency components with arbitrary phase angles.

The present invention utilizes both the amplitude sensitivity and the phase angle insensitivity of the human hearing mechanism to provide a speech privacy system. In this system a coded signal is derived from a speech wave by making the phase angles of the frequency components of the coded signal representative of the amplitudes of the corresponding frequency components of the speech wave, and by making the amplitudes of the frequency components of the coded signal uniform. Since neither the phase angles nor the uniform amplitudes of the frequency components of the coded signal convey information to the human hearing mechanism, the coded signal of this invention is unintelligible to unauthorized listeners. At an authorized receiver station, however, special decoding apparatus is provided in this invention for converting the phase angles of the components of the coded signal into the amplitudes of the corresponding components of a replica of the original speech wave.

In order to convert the phase angles of the coded signal components into the amplitudes of the replica components, it is necessary to transmit in addition to the coded signal a supplementary signal bearing a predetermined relationship to the coded signal: each component of the supplementary signal has the same uniform amplitude as the components of the coded signal, and the phase angle of each component of the supplementary signal differs from the phase angles of the components of the coded signal by an amount approximately proportional to the amplitude of a corresponding component of the original speech wave. Since each component of the supplementary signal is uniform in amplitude, the supplementary signal is also unintelligible to unauthorized listeners. At an authorized receiver station, there is subtracted from each uniform amplitude component of the coded signal a corresponding uniform amplitude component derived from the supplementary signal, thereby reconstructing a replica signal having frequency components with amplitudes that are approximately proportional to the amplitudes of corresponding frequency components of the original speech wave.

Two alternative arrangements embodying the principles of this invention are provided, one arrangement requiring a transmission channel having twice the bandwidth of the

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incoming speech wave, and the other arrangement requiring a transmission channel having the same bandwidth as the incoming speech wave. In the first arrangement, the uniform amplitude components of the supplementary signal correspond in number and frequency to both the components of the incoming speech wave and the components of the coded signal, hence the combined bandwidth of the coded and supplementary signals is twice the bandwidth of the incoming speech wave. Further, the phase angles of the frequency components of the supplementary signal are equal in magnitude but opposite in orientation to the phase angles of the corresponding components of the coded signal. By subtracting the supplementary signal from the coded signal at an authorized receiver station, there is obtained a corresponding group of frequency components having amplitudes approximately proportional to the amplitudes of the components of the speech wave.

In the second arrangement, the frequency components of the speech wave are given a predetermined uniform phase angle before the amplitudes of the speech frequency components are converted into phase angles of corresponding uniform amplitude components of a coded signal. The supplementary signal employed in the second arrangement has a single uniform amplitude frequency component with the same predetermined phase angle that the speech components are given before coding. The single component supplementary signal has a relatively narrow bandwidth, and by reducing the bandwidth of the coded signal by an amount on the order of the bandwidth of the supplementary signal, the combined bandwidth of the reduced bandwidth coded signal and the supplementary signal is made equal to the bandwidth of the speech wave. At an authorized receiver station there is derived from the single component of the supplementary signal a group of harmonics which correspond in frequency and number to the components of the reduced bandwidth coded signal, and which have the same predetermined phase angle and amplitude as the single component of the supplementary signal. By subtracting from the components of the coded signal the group of harmonics generated from the supplementary signal there is obtained a corresponding group of reconstructed frequency components having amplitudes proportional to the amplitudes of the components of the original speech wave.

The present invention also provides an additional privacy feature for use with the second alternative embodiment described above. At the transmitter station, the uniform phase angles that the speech frequency components are given before coding are changed by predetermined amounts according to a prearranged pattern known only to authorized listeners. Correspondingly, at the receiver station, these predetermined amounts are removed from the phase angles of the coded signal components before the harmonics derived from the supplementary signal are subtracted from the components of the coded signal.

This invention will be fully understood from the following detailed descriptions of illustrative embodiments thereof, taken in conjunction with the appended drawings, in which:

FIG. 1 is a block schematic diagram showing a speech privacy system embodying the principles of this invention;

FIG. 2 is a block schematic diagram showing an alternative speech privacy system embodying the principles of this invention;

FIGS. 3A, 3B, and 3C are vector diagrams of assistance in explaining certain features of this invention; and

FIGS. 4A and 4B are diagrams illustrating the frequency components of certain signals obtained in the apparatus shown in FIGS. 1 and 2.

Referring first to FIG. 1, an incoming speech wave S from source 10, which may be a conventional transducer such as a microphone for converting speech sounds into a corresponding electrical wave, is applied in parallel to a first signal path 1 and a second signal path 2. In signal path 1, the speech wave S is delivered to an amplitude flattening circuit 11 comprising a plurality of N parallel subpaths each of which contains a first bandpass filter 110, an amplitude limiting device, for example, an infinite clipper circuit 111, and a second bandpass filter 112, all connected in series. Bandpass filters 110-1 through 110- N of amplitude flattening circuit 11 separate the incoming wave S into its individual frequency components, denoted $s_1, s_2, \dots, s_N$, each frequency component being characterized by an amplitude and a phase angle determined by the nature of the particular speech sound represented by S at a given instant. Infinite clipper circuits 111-1 through 111- N make the amplitudes of the individual speech components uniform, and filters 112-1 through 112- N , which have pass bands identical with the pass bands of filters 110-1 through 110- N , remove unwanted distortion components from the uniform amplitude output signals of clippers 111-1 through 111- N . The uniform amplitude components, also referred to hereinafter as "flattened" components, which appear at the output terminals of filters 112-1 through 112- N are denoted $\bar{s}_1$ through $\bar{s}_N$ respectively, and these flattened components may be combined on common output bus 3 to form an output signal denoted $\bar{S}$. A graphical illustration of the operation of circuit 11 is provided by FIGS. 4A and 4B, in which the vertical lines of various amplitudes in FIG. 4A represent the harmonically related frequency components $s_1, s_2, \dots, s_N$ of a typical voiced speech sound before flattening, and the vertical lines of uniform amplitude in FIG. 4B represent the same components after flattening by circuit 11. By choosing the filters in flattening circuit 11 to have linear phase characteristics, the phase angles of the flattened components $\bar{s}_1$ through $\bar{s}_N$ are not affected by the amplitude flattening performed in circuit 11; that is, each flattened component $\bar{s}_1$ through $\bar{s}_N$ has the same phase angle as its original counterpart component s_1 through s_N . It is important at this point to distinguish between amplitude limiting applied to individual speech frequency components in this invention and amplitude limiting applied to the speech wave as a function of time. In the latter situation, the relative amplitudes of the speech frequency components are not affected by the amplitude limiting, and as a result, the speech wave remains highly intelligible despite the amplitude limiting; for example, see J. C. R. Licklider, "The Intelligibility of Amplitude-Dichotomized, Time-Quantized Speech Waves," volume 22, Journal of the Acoustical Society of America, page 820 (1950).

In signal path 2, the incoming speech wave S is passed through delay element 121, automatic gain control circuit 122, and 90° phase shifting circuit 123, all connected in series. Delay element 121 is designed in conventional fashion to compensate for the delay introduced by amplitude flattening circuit 11, while automatic gain control circuit 122 adjusts the amplitudes of the components of the incoming speech wave to lie within a desired amplitude range for reasons that will be explained in detail below. It is preferred that 90° phase shifting circuit 123 be a differentiating circuit; however, other 90° phase shifting circuits may be employed if desired. The output signal of 90° phase shifting circuit 123, denoted $\hat{S}$, therefore has individual frequency components that differ in phase angle and amplitude from the corresponding components of the output signal $\bar{S}$ of circuit 11: the individual components of $\hat{S}$, denoted $\hat{s}_1$ through $\hat{s}_N$, differ in phase by 90° from the corresponding components $\bar{s}_1$ through $\bar{s}_N$ of $\bar{S}$, and whereas the components of $\bar{S}$ are uniform in amplitude, the components $\hat{s}_1, \hat{s}_2, \dots, \hat{s}_N$ of $\hat{S}$ are

proportional to the amplitudes of the original speech components, the constant of proportionality being determined by the gain characteristic selected for automatic gain control circuit 122.

The relationships between the amplitudes and phase angles of the components of the original speech wave S and the two output signals $\hat{S}$ and $\bar{S}$ will be more easily understood by referring to FIG. 3A. In FIG. 3A, the n th component s_n of the original speech wave S is represented by a vector, also denoted s_n , this vector having a magnitude and phase angle respectively representative of the amplitude and phase angle of the corresponding speech component s_n . The uniform amplitude or flattened version of s_n is illustrated in FIG. 3A by the vector labeled $\bar{s}_n$, in which it is observed that $\bar{s}_n$ has the same phase angle but a smaller magnitude than the vector s_n . Also, in FIG. 3A, the vector labeled $\hat{s}_n$ represents the n th component $\hat{s}_n$ of the output signal $\hat{S}$ appearing at the output terminal of 90° phase shifter 123, in which it is noted that the vector $\hat{s}_n$ is perpendicular to both of the vectors s_n and $\bar{s}_n$. In addition, the vector labeled $-\hat{s}_n$ in FIG. 3A illustrates the oppositely directed 90° phase shifted component, $-\hat{s}_n$, while the resultant vectors labeled $\sigma_{1n} = (\bar{s}_n + \hat{s}_n)$, $\sigma_{2n} = (\bar{s}_n - \hat{s}_n)$, illustrate the components which are obtained by respectively combining the components of the output signal $\bar{S}$ with the components of $\hat{S}$ and $-\hat{S}$.

From FIG. 3A it is evident that the phase angle differences φ_n and $-\varphi_n$ between the phase angles of the respective resultant vectors σ_{1n} , σ_{2n} , and the phase angle of $\bar{s}_n$ are proportional to the magnitude of $\hat{s}_n$, so that by either adding or subtracting the 90° phase shifted components of the output signal $\hat{S}$ from the corresponding uniform amplitude components of the output signal $\bar{S}$, the amplitudes of the components of S are converted into phase angles of the components of respective sum and difference signals,

$$\sigma_{1n} = (\bar{s}_n + \hat{s}_n) \quad (1a)$$

and

$$\sigma_{2n} = (\bar{s}_n - \hat{s}_n) \quad (1b)$$

Since the amplitude of a 90° phase shifted component $\hat{s}_n$ is proportional to the amplitude of a corresponding component s_n of the original speech wave, as explained above, the phase angle φ_n , $-\varphi_n$ of the respective sum and difference components σ_{1n} , σ_{2n} , measured with respect to the phase angle of $\bar{s}_n$, are also representative of the amplitude of the original speech component s_n . Since the human hearing mechanism is responsive primarily to the amplitudes but not to the phase angles of the components of a speech signal, Equations 1a and 1b illustrate one of the principles of this invention for converting the information contained in speech component amplitudes into a form that cannot be detected by the unaided human hearing mechanism.

Apparatus for implementing this principle is illustrated in FIG. 1, where there is provided an adding circuit 13 and a subtracting circuit 14 for combining the components of the output signals of circuit 11 and circuit 123 in accordance with Equations 1a and 1b. The output signal $\bar{S}$ of amplitude flattening circuit 11 is delivered simultaneously to one input terminal of adding circuit 13 and to the minuend terminal of a conventional subtracting circuit 14, while the output signal $\hat{S}$ of the 90° phase shifting circuit 123 is applied simultaneously to a second input terminal of adding circuit 13 and to the subtrahend terminal of subtracting circuit 14. Adding circuit 13 develops from the individual components $\bar{s}_1, \dots, \bar{s}_N$ of $\bar{S}$ and the individual components $\hat{s}_1, \dots, \hat{s}_N$ of $\hat{S}$ a corresponding group of sum components $\sigma_{11}, \dots, \sigma_{1N}$, while subtracting circuit 14 develops a corresponding group of difference components $\sigma_{21}, \dots, \sigma_{2N}$. The out-

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put signals defined by the sum and difference components developed by adding circuit 13 and subtracting circuit 14 may be respectively denoted $\Sigma_1 = (\bar{S} + \hat{S})$ and $\Sigma_2 = (\bar{S} - \hat{S})$, and as previously noted, the effect of adding and subtracting the components of $\bar{S}$ and $\hat{S}$ in this manner is illustrated in FIG. 3A by the adding and subtracting of the vectors $\bar{s}_n$ and $\hat{s}_n$ respectively representing individual components of $\bar{S}$ and $\hat{S}$ to produce the two resultant vectors σ_{1n} and σ_{2n} .

However, it is observed in FIG. 3A that the combining of $\bar{s}_n$ and $\hat{s}_n$ in accordance with Equations 1a and 1b not only effects a conversion of amplitudes into phase angles but also causes the amplitudes of the combined components to be proportional to the amplitudes of the original speech components, since the magnitudes $|\sigma_{1n}|$, $|\sigma_{2n}|$, of the combined or resultant vectors in FIG. 3A evidently reflect the magnitude of the original speech component represented by the vector $\hat{s}_n$. In order to achieve the desired privacy, it is therefore necessary to remove from the amplitudes of these combined components any information regarding the original speech component amplitudes. Further, the removal of amplitude information from the combined components must not affect the phase angles of the combined components in order to avoid any impairment of the speech information contained in the phase angles of these components. This is accomplished by applying each of the combined signals σ_1 and σ_2 to respective amplitude flattening circuits 15 and 16, which may be identical in construction with amplitude flattening circuit 11, thereby making each component of the combined signals uniform in amplitude without altering the phase angles of these components. Each of the output signals obtained at the output terminal of amplitude flattening circuits 15 and 16 is respectively referred to hereinafter as a coded signal and a supplementary signal and is respectively denoted

$$\bar{\Sigma}_1 = (\bar{S} + \hat{S}) \text{ and } \bar{\Sigma}_2 = (\bar{S} - \hat{S})$$

As observed in FIG. 3A, the magnitude of each of the corresponding flattened resultant vectors, denoted

$$\bar{\sigma}_{1n} = (\bar{s}_n + \hat{s}_n) \text{ and } \bar{\sigma}_{2n} = (\bar{s}_n - \hat{s}_n)$$

differs from the magnitude of each of the respective unflattened resultant vectors σ_{1n} and σ_{2n} , but the flattened resultant vectors $\bar{\sigma}_{1n}$ and $\bar{\sigma}_{2n}$ have the same phase angles as the unflattened resultant vectors σ_{1n} and σ_{2n} .

The coded signal $\bar{\Sigma}_1$ and the supplementary signal $\bar{\Sigma}_2$ from the two amplitude flattening circuits 15 and 16 may be delivered by means of a suitable transmission medium, indicated by broken lines, to a receiver station. Speech transmitted in this fashion is relatively secure from the standpoint of privacy, for the amplitude information conveyed by the phase angles of the individual components of the coded and supplementary signals cannot be detected by unauthorized listeners using conventional speech detection apparatus. However, with the special processing apparatus described below, intelligible speech may be recovered from the phase angles of the transmitted signals at an authorized receiver station.

As shown at the receiver station in FIG. 1, intelligible speech is obtained at an authorized receiver station by subtracting the supplementary signal components from the coded signal components so that the phase angles of the components of the coded signal are converted into the amplitudes of the components of a reconstructed signal. For the reasons given below, the amplitudes of the components of the reconstructed signal closely follow the amplitudes of the components of the original speech signal, hence the reconstructed signal is a highly intelligible replica of the original speech wave. By applying the reconstructed signal to a suitable reproducer 18, for example, a loudspeaker of conventional design, intelligible speech sounds may be obtained.

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The principle underlying the conversion of phase angles into amplitudes by subtracting the components of the supplementary signal from the components of the coded signal to obtain an intelligible replica of the original speech signal will be understood from the following considerations. Referring back to FIG. 3A, the phase angle φ_n may be expressed in terms of the magnitude of the vector $\hat{s}_n$ representing the 90° phase shifted components and the vector $\bar{s}_n$ representing the flattened components,

$$\varphi_n = \tan^{-1} \left(\frac{|\hat{s}_n|}{|\bar{s}_n|} \right) \quad (2)$$

By subtracting $\bar{\sigma}_{2n}$ from $\bar{\sigma}_{1n}$ there is obtained a difference vector, denoted $\Delta\sigma_n = (\bar{\sigma}_{1n} - \bar{\sigma}_{2n})$, which is perpendicular to the flattened vector $\bar{s}_n$, so that the magnitude of the difference vector, denoted $|\Delta\sigma_n|$, is exactly divided in two by the flattened vector $\bar{s}_n$. The sine of the phase angle φ_n may therefore be written as

$$\sin \varphi = \frac{1/2 |\Delta\sigma_n|}{|\bar{\sigma}_{1n}|} \quad (3)$$

By constructing amplitude flattening circuits 11, 15, and 16 in the apparatus of FIG. 1 so that all of the flattened components have the same uniform amplitudes, that is,

$$|\bar{s}_n| = |\sigma_{1n}| = |\sigma_{2n}| \quad (4)$$

Equation 3 may be rewritten

$$|\Delta\sigma_n| = 2 \cdot |\bar{s}_n| \sin \left[\tan^{-1} \left(\frac{|\hat{s}_n|}{|\bar{s}_n|} \right) \right] \quad (5a)$$

Using the well-known trigonometric identity,

$$\sin x = \frac{\tan x}{\sqrt{1 + \tan^2 x}} \quad (5b)$$

the magnitude of the difference vector, $|\Delta\sigma|$, in Equation 5a may be expressed as

$$\begin{aligned} |\Delta\sigma_n| &= 2 \cdot |\bar{s}_n| \cdot \frac{\left(\frac{|\hat{s}_n|}{|\bar{s}_n|} \right)}{\sqrt{1 + \left(\frac{|\hat{s}_n|}{|\bar{s}_n|} \right)^2}} \\ &= 2 \cdot \frac{|\hat{s}_n|}{\sqrt{1 + \left(\frac{|\hat{s}_n|}{|\bar{s}_n|} \right)^2}} \end{aligned} \quad (6)$$

An expression for the difference vector itself may be obtained by removing the absolute value sign in the numerator on the right-hand side of Equation 6, hence

$$\Delta\sigma_n = 2 \frac{\hat{s}_n}{\sqrt{1 + \left(\frac{|\hat{s}_n|}{|\bar{s}_n|} \right)^2}} \quad (7)$$

From Equation 6 it is evident that the magnitude of each of the difference components of the reconstructed signal obtained at an authorized receiver of this invention is approximately proportional to the magnitude of the corresponding component $\hat{s}_n$ of the 90° phase shifted signal, $\hat{S}$, provided that the value of the denominator factor on the right-hand side of Equation 6 remains relatively constant. By constructing both automatic gain control circuit 122 and amplitude flattening circuit 11 to adjust appropriately the respective amplitudes of $\hat{s}_n$ and $\bar{s}_n$, the term

$$\left(\frac{|\hat{s}_n|}{|\bar{s}_n|} \right)$$

in Equation 6 may be controlled so that the denominator factor in Equation 6 does not depart widely from a predetermined constant value. Since the amplitude of a 90° phase shifted component $\hat{s}_n$ is proportional to the amplitude of the corresponding component s_n of the original speech wave, Equation 6 specifies that the amplitude of the corresponding difference component of the reconstructed signal is also approximately proportional to the amplitude of the original speech component, thereby dem-

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onstrating that the subtraction of signals at the receiver station of the apparatus in FIG. 1 converts phase angles into amplitudes to reconstruct a replica of the original speech wave. From Equation 7, however, it is observed that the phase angles of the reconstructed components differ by 90° from the phase angles of the corresponding components of the speech wave.

It is observed in the system shown in FIG. 1 that the coded signal and the supplementary signal respectively obtained from circuits 15 and 16 and transmitted to a receiver station each has a bandwidth equal to the bandwidth of the incoming speech wave, since the components of the coded and supplementary signals are identical in number and frequency with the components of the speech wave. Hence it is necessary to employ either a relatively wide bandwidth transmission channel or two separate channels each wide enough to accommodate one of the output signals of circuits 15 and 16. However, it may be desirable to have a privacy system embodying the principles of this invention which employs a transmission channel having a bandwidth on the order of the bandwidth of the incoming speech wave. Such a privacy system is illustrated in FIG. 2. Turning now to FIG. 2, at the transmitter station of the system an incoming speech wave S from source 10 is applied to a complete channel vocoder 21, which may be of the type described in H. W. Dudley Patent 2,151,091, issued Mar. 21, 1939. Within vocoder 21, the incoming speech wave S is delivered in parallel to an analyzer 212 and a pitch detector 213. Analyzer 212 derives from S a plurality of reduced bandwidth control signals representative of the amplitudes of the individual frequency components of S, while pitch detector 213 derives from S a train of pulses having a fundamental frequency equal to that of the fundamental component of voiced portions of the speech wave. The control signals from analyzer 212 are applied to synthesizer 214, together with an excitation signal derived by excitation generator 215 from the train of pulses from detector 213, thereby developing at the output terminal of vocoder 21 a synthesized speech wave S_0 that is a highly intelligible replica of the original speech signal. The synthesized speech wave S_0 is characterized by frequency components having amplitudes proportional to the amplitudes of corresponding components of the original speech wave, but the phase angles of the synthesized speech components are uniform and bear no particular relationship to the individual phase angles of the corresponding components of the original speech wave. For example, the phase angles of the synthesized components may all be made equal to zero, if desired, by suitably adjusting the phase of the excitation signal from generator 215.

The synthesized speech signal from vocoder 21 is applied through switch P1 to parallel signal paths 1 and 2, signal paths 1 and 2 respectively including elements identical to those in signal paths 1 and 2 in the apparatus of FIG. 1. Thus signal path 1 includes an amplitude flattening circuit 11, while signal path 2 includes a delay element 121 followed by an automatic gain control circuit 122 and a 90° phase shifting circuit 123 connected in series. The output signal of amplitude flattening circuit 11 may be denoted by $\bar{S}_0$, with individually flattened components denoted $\bar{s}_{01}, \dots, \bar{s}_{0N}$, while the output signal of circuit 123 may be denoted by $\hat{S}_0$, with individually phase shifted components denoted $\hat{s}_0, \dots, \hat{s}_N$. The individual components of the two signals, $\bar{S}_0$ and $\hat{S}_0$, are combined in a conventional adding circuit 234 to produce a resultant signal denoted $\Sigma_0 = \bar{S}_0 + \hat{S}_0$, and having individual sum components denoted $\sigma_{01}, \dots, \sigma_{0N}$, where

$$\sigma_{0n} = \bar{s}_{0n} + \hat{s}_{0n}$$

By combining the individual components of $\bar{S}_0$ and $\hat{S}_0$ in adding circuit 234, the amplitudes of the components of the synthesized signal S_0 are converted into phase angles of the corresponding components of the combined signal

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Σ_0 in the same manner that the combining of the signals $\bar{S}$ and $\hat{S}$ in the apparatus of FIG. 1 converted amplitudes into phase angles. Similarly, the combined signal Σ_0 is passed through amplitude flattening circuit 23 in the apparatus of FIG. 2 in order to remove from the amplitudes of the sum components $\sigma_{01}, \dots, \sigma_{0N}$ all information regarding the amplitudes of the original speech components, just as the combined signals Σ_1 and Σ_2 in the apparatus of FIG. 1 were passed through spectrum flattening circuits 15 and 16, respectively.

The relationships between the synthesized speech signal S_0 and the two output signals $\bar{S}_0$ and $\hat{S}_0$ are illustrated graphically in FIG. 3B, where the vectors labeled s_{0n} , $\bar{s}_{0n}$, and $\hat{s}_{0n}$ respectively represent individual n^{th} components of the signals S_0 , $\bar{S}_0$, $\hat{S}_0$. It is observed that the vector $\hat{s}_{0n}$ is perpendicular to the vector $\bar{s}_{0n}$, so that by adding $\hat{s}_{0n}$ and $\bar{s}_{0n}$, there is obtained a sum vector

$$\sigma_{0n} = (\hat{s}_{0n} + \bar{s}_{0n})$$

which differs in phase angle from s_{0n} and $\bar{s}_{0n}$ by an amount φ_n that is proportional to the magnitude of $\hat{s}_{0n}$ and therefore is also proportional to the amplitude of the n^{th} original speech component represented by s_{0n} . Hence

by combining $\bar{S}_0$ and $\hat{S}_0$ in adder 234 of FIG. 2 there is obtained a sum signal Σ_0 with individual sum components having phase angles proportional to the amplitudes of the corresponding components of the original speech wave.

However, the magnitude of vector σ_{0n} in FIG. 3B is also proportional to the magnitude of vector s_{0n} , so that for purposes of privacy it is necessary to remove from the amplitudes of the components of the combined signal Σ_0 represented by the vector s_{0n} in FIG. 3B all traces of the amplitudes of the original speech components represented by the vector s_{0n} . This is accomplished in the apparatus of FIG. 2 by passing the combined signal $\Sigma_0 = \bar{S}_0 + \hat{S}_0$ from adder 234 through an amplitude flattening circuit 24, which may be identical in construction with circuit

11, thereby obtaining a coded output signal $\bar{\Sigma}_0 = (\bar{S}_0 + \hat{S}_0)$. The components of the coded output signal $\bar{\Sigma}_0$ derived by circuit 24 are illustrated graphically in FIG. 3B by the uniform magnitude vector denoted $\bar{\sigma}_{0n}$, and $\bar{\Sigma}_0$ is unintelligible to unauthorized listeners since only the phase angles of the uniform amplitude components of Σ_0 contain speech information. In order to decode the signal, $\bar{\Sigma}_0$, however, it is necessary to transmit additional information which will enable the phase angles of the uniform amplitude components of $\bar{\Sigma}_0$ to be converted into the amplitudes of the components of a reconstructed, intelligible speech signal at an authorized receiver station. This additional information may be provided by deriving a supplementary signal having the amplitude and phase angle information contained in the train of pulses from pitch detector 213 of vocoder 21.

As shown in FIG. 2, a supplementary signal is obtained by applying the train of pulses from detector 213 to a tracking filter 25, which is constructed to track the fundamental frequency component, denoted f_0 , of the train of pulses. A suitable design for tracking filter 25 is described in C. B. H. Feldman et al. Patent 2,859,405, issued Nov. 4, 1958. The output signal of filter 25 is characterized by a single frequency component having the same uniform amplitude as the components of the coded signal; this may be accomplished, for example, by constructing flattening circuits 11 and 23 to make the uniform amplitudes of the components of $\bar{S}_0$ and $\bar{\Sigma}_0$ equal not only to each other but also to the amplitude of the fundamental component of the train of pulses from detector 213. Further, since the synthesized output signal S_0 of vocoder 21 is reconstructed from the train of pulses produced by detector 213, the output signal of filter 25 has the same uniform phase angle as the components of S_0 and $\bar{S}_0$.

The suitability of the output signal of filter 25 as a supplementary signal for decoding the coded signal $\bar{S}_0$ is illustrated in FIG. 3B. Referring to FIG. 3B, it is observed that by subtracting a vector $\bar{s}_{0n}$ representing the component of the supplementary signal from the vector $\bar{\sigma}_{0n}$ representing a component of the coded signal there is obtained a difference vector $\Delta s_n = (\bar{\sigma}_{0n} - \bar{s}_{0n})$, and it is shown below that the magnitude of this difference vector, $|\Delta s_n|$, is proportional to the magnitude of the original vector s_{0n} . Hence by transmitting the output signal f_0 of filter 25 together with the coded signal $\bar{S}_0$ from circuit 23, an intelligible replica of the original speech signal may be reconstructed at an authorized receiver station by subtracting from the components of the coded signal a corresponding group of components derived from f_0 , where the derived components all have uniform amplitudes equal to the uniform amplitudes of the components of the coded signal, $\bar{S}_0$, and uniform phase angles equal to the uniform phase angles of the components of the synthesized signal, S_0 .

In FIG. 3B, by subtracting the vector $\bar{s}_{0n}$ from the vector $\bar{\sigma}_{0n}$, there is obtained a difference vector denoted Δs_n , where

$$\Delta s_n = \bar{\sigma}_{0n} - \bar{s}_{0n}$$

It is observed in FIG. 3B that the phase angle φ_n may be expressed in terms of the magnitudes of vectors s_{0n} and $\bar{s}_{0n}$, exactly as in the case of the phase angle φ_n in FIG. 3A, that is,

$$\varphi_n = \tan^{-1} \left(\frac{|\hat{s}_{0n}|}{|\bar{s}_{0n}|} \right) \quad (8)$$

Since $\bar{s}_{0n}$ and $\bar{\sigma}_{0n}$ are equal in magnitude, a line that bisects φ_n also bisects the difference vector Δs_n , hence the sine of the angle

$$\frac{1}{2} \varphi_n$$

may be expressed as

$$\sin \frac{1}{2} \varphi_n = \frac{\frac{1}{2} |\Delta s_n|}{|\bar{s}_{0n}|} \quad (9a)$$

or

$$|\Delta s_n| = 2 |\bar{s}_{0n}| \sin \frac{1}{2} \varphi_n \quad (9b)$$

Substituting Equations 5b and 8 into Equation 9b, the magnitude of the difference vector may be written

$$|\Delta s_n| = \frac{|\hat{s}_{0n}|}{\sqrt{1 + \left(\frac{|\hat{s}_{0n}|}{|\bar{s}_{0n}|} \right)^2}} \quad (10)$$

and by removing the absolute value sign in the numerator of the expression on the right-hand side of Equation 10, the difference vector itself may be written

$$\Delta s_n = \frac{\hat{s}_{0n}}{\sqrt{1 + \left(\frac{|\hat{s}_{0n}|}{|\bar{s}_{0n}|} \right)^2}} \quad (11)$$

Since the magnitude of the 90° phase shifted vector $\hat{s}_{0n}$ is proportional to the magnitude of the original vector s_{0n} , it is evident from Equation 10 that the magnitude of the difference vector Δs_n is approximately proportional to the magnitude of the original vector s_{0n} , provided that the value of the denominator in the term on the right-hand side of Equation 10 remains relatively constant. Hence an intelligible replica of the original speech wave may be reconstructed by subtracting from each of the uniform amplitude components of the coded signal a corresponding component having the same uniform amplitude and a phase angle equal to the uniform phase angle of the components of the synthesized speech signal.

However, in order to obtain reconstructed components that closely approximate the amplitudes of the corresponding components of the original speech wave, it is necessary to select the gain characteristics of automatic gain control circuit 122 and amplitude flattening circuit 11 so that the ratio of the amplitudes of corresponding components of $\hat{S}_0$ and $\bar{S}_0$ is relatively constant as required by the denominator term,

$$\frac{|\hat{s}_{0n}|}{|\bar{s}_{0n}|}$$

in Equation 10.

Turning back to FIG. 2, the supplementary signal f_0 from filter 25 is transmitted to a receiver station together with the coded signal $\bar{S}_0$ of circuit 23; however, in order to accommodate both f_0 and $\bar{S}_0$ within a single transmission channel having a bandwidth equal to that of the incoming speech wave S , an appropriate low frequency portion of the coded signal $\bar{S}_0$ is removed by high pass filter 24 before transmitting $\bar{S}_0$ together with f_0 .

At an authorized receiver station, the supplementary signal f_0 is recovered from the transmitted signal by low pass filter 26, and from f_0 to a suitable harmonic generator 27, for example, a free running multivibrator, generates harmonics of the frequency component of f_0 which have the same amplitude and phase angle as the frequency component of f_0 ; that is, the harmonics generated by generator 27 correspond in frequency to, and have the same uniform amplitude as, the components of the coded signal $\bar{S}_0$, and have the same uniform phase angle as the components of the synthesized signal S_0 from vocoder 21. In order to take into account equipment variations and inaccuracies that may cause the phase angles of the components of the harmonics from generator 27 to differ from the phase angles of the components of S_0 , there is provided a variable delay element 28 which may be adjusted by a user at the receiver station to restore the phase angles of the components of the train of pulses to their proper value.

The coded signal is passed through high pass filter 30 to remove the f_0 signal, and switch P2, which is set to correspond to the setting of switch P1 at the transmitter station, delivers the $\bar{S}_0$ signal to the minuend terminal of conventional subtracting circuit 29. Correspondingly, delay element 28 delivers the harmonics from generator 27 to the subtrahend terminal of subtracting circuit 29, and circuit 29 develops at its output terminal a signal that is an intelligible replica of the incoming speech signal in that the amplitudes of the components of the replica signal closely follow the amplitudes of the corresponding components of the incoming speech signal. A reproducer 20, which may be a conventional loudspeaker, converts the replica signal from circuit 29 into audible, intelligible speech sounds.

If desired, the system shown in FIG. 2 may be adapted to provide an additional degree of privacy by setting switches P1 and P2 to include phase transformation circuit 22a and inverse phase transformation circuit 22b in the system, where circuits 22a and 22b may be all-pass networks constructed according to the criteria set forth in S. Darlington, "Realization of a Constant Phase Difference," volume 29, Bell System Technical Journal, page 94 (1950). Phase transformation circuit 22a at the transmitter station is constructed to change the phase angles of the individual components of the synthesized speech signal S_0 by various amounts before coding according to a predetermined schedule known only to legitimate receivers of the transmitted signal. Correspondingly, inverse phase transformation circuit 22b at the receiver station is constructed to remove from the components of the coded signal the phase angle changes introduced at the transmitter station, following which the components of

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the coded signal are decoded in the manner previously described.

The effect of circuits 22a and 22b is illustrated graphically in FIG. 3C. A change in the phase angle of the vector s_{0n} by a predetermined amount ρ corresponds to the passage of the components of S_0 through circuit 22a to produce a new signal, S_r , whose n th component, s_{rn} , has the same magnitude as the corresponding incoming component, s_{0n} , but a different phase angle. Amplitude flattening circuit 11 in subpath 1 makes the components of S_r uniform in amplitude to derive an amplitude flattened signal $\bar{S}_r$, as indicated by the vector $\bar{s}_{rn}$ in FIG. 3C, while the components of the output signal $\hat{S}_r$ of subpath 2 are shifted in phase by 90° relative to the components of S_r , as indicated by the vector $\hat{s}_{rn}$ in FIG. 3C. By combining the components of $\bar{S}_r$ and $\hat{S}_r$ in adder 234, there is obtained a sum signal $\Sigma_r = (\bar{S}_r + \hat{S}_r)$, represented by vector $\sigma_r = (\bar{s}_{rn} + \hat{s}_{rn})$ in FIG. 3C, and amplitude flattening circuit 23 makes the components of the sum signal Σ_r uniform in amplitude, as shown by vector $\bar{\sigma}_r$ in FIG. 3C. At the receiver station, circuit 22b removes from the phase angles of the components of $\bar{\Sigma}_r$ the amounts of phase angle change introduced by circuit 22a at the transmitter station as indicated by the phase angle difference ρ between the vector $\bar{\sigma}_r$ and the vector $\bar{\sigma}_{r-\rho}$ in FIG. 3C, thereby enabling subtracting circuit 29 to convert the phase angles of the components of the coded signal into the amplitudes of the components of a replica of the original speech signal.

It is to be understood that applications of the principles of this invention are not limited to the specific examples described above, but may include other communication systems in which it is desired to transmit speech information with privacy. In addition, it is to be understood that the above-described embodiments are merely illustrative of the numerous arrangements that may be devised for the principles of this invention by those skilled in the art without departing from the spirit and scope of this invention.

What is claimed is:

1. A speech privacy system that comprises a transmitter station including a source of an incoming speech wave having a plurality of frequency components, wherein each of said frequency components is characterized by an amplitude and a phase angle, means for delivering said speech wave in parallel to a first subpath in which there is derived from each of said frequency components a corresponding uniform amplitude component and to a second subpath in which there is derived from each of said frequency components a corresponding 90° phase shifted component, means for additively combining each uniform amplitude component with its corresponding 90° phase shifted component to obtain a corresponding plurality of sum components, each sum component being characterized by an amplitude and a phase angle proportional to the amplitude of a corresponding frequency component of said speech wave, means for subtracting each 90° phase shifted component from its corresponding uniform amplitude component to obtain a corresponding plurality of difference components, each difference component being characterized by an amplitude and a phase angle proportional to a corresponding frequency component of said speech wave, means for making uniform the amplitude of each of said plurality of sum components to derive a corresponding plurality of uniform amplitude sum components, means for making uniform the amplitude of each of

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said plurality of difference components to derive a corresponding plurality of uniform amplitude difference components,

means for transmitting said plurality of uniform amplitude sum components and said plurality of uniform amplitude difference components to a receiver station, and

at said receiver station,

means for subtracting each of said uniform amplitude difference components from a corresponding one of said uniform amplitude sum components to obtain a plurality of frequency components having amplitudes representative of the amplitudes of corresponding frequency components of said incoming speech wave.

2. A speech privacy system that comprises

a transmitter station including

means for converting human speech sounds into an incoming speech wave characterized by a plurality of frequency components, each of said frequency components having a certain amplitude and a certain phase angle determined by the nature of said speech sounds at a given instant,

means for deriving from said speech wave a pair of first and second coded signals, wherein each of said coded signals is characterized by a plurality of uniform amplitude frequency components in one-to-one correspondence with each other and with said frequency components of said speech wave, and wherein the difference in phase angle between each of said frequency components of said first and second coded signals and said corresponding frequency component of said speech wave is proportional to the amplitude of said corresponding frequency component of said speech wave,

means for transmitting said pair of first and second coded signals to a receiver station, and

at said receiver station,

means for reconstructing a replica of said speech wave from said pair of first and second coded signals.

3. Apparatus for coding a message signal by converting the amplitudes of the frequency components of said message signal into the phase angles of corresponding frequency components of a coded signal which comprises

amplitude flattening means for making the amplitudes of said frequency components of said message signal equal to a predetermined uniform value to obtain corresponding uniform amplitude frequency components,

phase shifting means in parallel with said amplitude flattening means for shifting the phase angle of each of said frequency components of said message signal by 90° to obtain corresponding phase shifted frequency components having amplitudes proportional to the amplitudes of corresponding frequency components of said message signal, and

means for additively combining each of said uniform amplitude frequency components with each of said corresponding phase shifted components to form said coded signal.

4. The method of reconstructing intelligible speech sounds from a pair of first and second coded signals in which said first coded signal is characterized by a plurality of frequency components each having the same predetermined uniform amplitude and each having a phase angle proportional to the amplitude of a corresponding frequency component of an original speech wave, and in which said second coded signal is characterized by a corresponding plurality of frequency components each having the same predetermined uniform amplitude as said components of said first coded signal, and each having a phase angle equal in magnitude but opposite in sign to the phase angle of the corresponding components of said first coded signal, which consists of the steps of:

subtracting the frequency components of said second

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coded signal from the corresponding frequency components of said first coded signal to obtain a reconstructed signal characterized by a plurality of frequency components having amplitudes representative of the amplitudes of corresponding components of said speech wave, and
 converting said reconstructed signal into intelligible speech sounds.

5. A speech privacy system that comprises
 a transmitter station including
 first transducer means for converting speech sounds into a corresponding speech wave characterized by a plurality of frequency components having various amplitudes and phase angles,
 first and second subpaths connected in parallel to said transducer means,
 said first subpath including a first amplitude flattening means for obtaining from said plurality of frequency components of said speech wave a corresponding plurality of uniform amplitude frequency components having the same phase angles as said corresponding frequency components of said speech wave,
 said second subpath including a delay element, an automatic gain control circuit, and a 90° phase shifting circuit connected in tandem for obtaining from said plurality of frequency components of said speech wave a corresponding plurality of 90° phase shifted frequency components having amplitudes proportional to the amplitudes of said corresponding frequency components of said speech wave,
 adding means for combining each of said uniform amplitude frequency components with a corresponding one of said 90° phase shifted frequency components to obtain a corresponding plurality of sum components,
 first subtracting means in parallel with said adding means for subtracting each of said 90° phase shifted components from a corresponding one of said uniform amplitude frequency components to derive a corresponding plurality of difference components,
 a second amplitude flattening means following said adding means for deriving from said plurality of sum components a corresponding plurality of uniform amplitude sum components that define a coded signal,
 a third amplitude flattening means following said subtracting means for deriving from said plurality of difference components a corresponding plurality of uniform amplitude difference components that define a supplementary signal,
 means for transmitting said coded signal and said supplementary signal to a receiver station, and
 at said receiver station,
 second subtracting means for subtracting said uniform amplitude difference components of said supplementary signal from said corresponding uniform amplitude sum components of said coded signal to derive a plurality of synthesized frequency components in one-to-one correspondence with said plurality of frequency components of said speech wave, wherein the amplitudes of said synthesized frequency components are representative of the amplitudes of corresponding frequency components of said speech wave, and
 second transducer means for reproducing speech sounds from said plurality of synthesized frequency components.

6. Apparatus as defined in claim 5 wherein said first amplitude flattening means comprises
 an input point,
 a first plurality of parallel bandpass filters each provided with an input terminal and an output terminal,
 means for connecting the input terminal of each of said first plurality of bandpass filters to said input point,

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a plurality of parallel amplitude limiting means in one-to-one correspondence with said first plurality of parallel bandpass filters, each of said amplitude limiting means being provided with an input terminal and an output terminal,
 means for connecting the output terminal of each of said bandpass filters to the input terminal of the corresponding one of said plurality of amplitude limiting means,
 a second plurality of parallel bandpass filters in one-to-one correspondence with said plurality of amplitude limiting means, each of said second plurality of bandpass filters being provided with an input terminal and an output terminal,
 means for connecting the output terminal of each of said amplitude limiting means to the input terminal of the corresponding one of said second plurality of bandpass filters,
 an output point, and
 means for connecting the output terminal of each of said second plurality of bandpass filters to said output point.

7. A speech privacy system that comprises
 a source of an incoming speech wave characterized by a plurality of frequency components having various amplitudes and phase angles,
 vocoder means for deriving from said speech wave a synthesized speech wave and a pitch signal,
 wherein said pitch signal is characterized by a plurality of frequency components all having a predetermined uniform amplitude and a predetermined uniform phase angle, and
 wherein said synthesized speech wave is characterized by a plurality of synthesized frequency components in one-to-one correspondence with said plurality of frequency components of said speech wave and each of said synthesized frequency components has a predetermined uniform phase angle equal to said predetermined uniform phase angle of said pitch signal components and an amplitude proportional to the amplitude of a corresponding one of said frequency components of said speech wave, and
 first and second subpaths connected in parallel to said vocoder means,
 said first subpath including a first amplitude flattening means for making the amplitudes of said plurality of synthesized frequency components equal to said predetermined uniform amplitude of said frequency components of said pitch signal to obtain a plurality of uniform amplitude synthesized frequency components,
 said second subpath including a delay element, an automatic gain control circuit, and a 90° phase shifting circuit connected in tandem for deriving from said plurality of frequency components of said synthesized speech wave a corresponding plurality of 90° phase shifted synthesized frequency components having amplitudes proportional to the amplitudes of corresponding frequency components of said speech wave,
 adding means for combining each of said plurality of uniform amplitude synthesized frequency components with a corresponding one of said 90° phase shifted synthesized frequency components to obtain a corresponding plurality of sum components,
 filter means supplied with said sum signal for passing selected ones of said sum components to form a reduced bandwidth coded signal,
 tracking filter means supplied with said pitch signal for selecting one of said frequency components of said pitch signal to serve as a supplementary signal,
 means for transmitting said reduced bandwidth coded signal and said supplementary signal to a receiver station, and
 at said receiver station,

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means for generating from said supplementary signal a plurality of harmonics of said frequency component of said supplementary signal, wherein said plurality of harmonics have amplitudes and phase angles respectively equal to said predetermined uniform amplitude and predetermined uniform phase angle of said frequency components of said pitch signal, and wherein said plurality of harmonics are in one-to-one correspondence with said selected sum components of said reduced bandwidth coded signal, 5

means for subtracting each of said plurality of harmonics from a corresponding one of said selected frequency components of said reduced bandwidth coded signal to obtain a plurality of reconstructed frequency components having amplitudes that closely follow the amplitudes of corresponding selected frequency components of said speech wave. 10

8. A speech privacy system that comprises a source of an incoming speech wave characterized by a plurality of frequency components having various amplitudes and phase angles, 20

channel vocoder means for deriving from said speech wave a synthesized speech wave and a pitch signal, wherein said pitch signal is characterized by a plurality of frequency components all having a predetermined uniform amplitude and a predetermined uniform phase angle, and 25

wherein said synthesized speech wave is characterized by a plurality of synthesized frequency components in one-to-one correspondence with said plurality of frequency components of said speech wave and each of said synthesized frequency components has a predetermined uniform phase angle equal to said predetermined uniform phase angle of said pitch signal components and an amplitude proportional to the amplitude of a corresponding one of said frequency components of said speech wave, and 30

phase transformation means for changing said predetermined uniform phase angle of each of said synthesized frequency components by a preselected amount to obtain a corresponding plurality of phase transformed frequency components, 40

first and second subpaths connected to said phase transformation means, said first subpath including a first amplitude flattening means for making the amplitudes of said plurality of phase transformed frequency components equal to said predetermined uniform amplitude of said frequency components of said pitch signal to obtain a corresponding plurality of uniform amplitude, phase transformed frequency components, 50

said second subpath including a delay element, an automatic gain control circuit, and 90° phase shifting circuit connected in tandem for deriving from said plurality of phase transformed frequency components a corresponding plurality of 90° phase shifted frequency components, 55

adding means for combining each of said plurality of uniform amplitude, phase transformed frequency

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components with a corresponding one of said 90° phase shifted frequency components to obtain a coded signal characterized by a corresponding plurality of sum components,

filter means supplied with said coded signal for passing selected ones of said plurality of sum components to derive a reduced bandwidth coded signal,

tracking filter means supplied with said pitch signal for selecting a predetermined one of said frequency components of said pitch signal to serve as a supplementary signal,

means for transmitting said reduced bandwidth coded signal and said supplementary signal to a receiver station, and

at said receiver station,

second filter means for separating said reduced bandwidth coded signal from said supplementary signal,

inverse phase transformation means supplied with said reduced bandwidth coded signal for changing the phase angles of each of said selected sum components by an amount equal in magnitude but opposite in sign to said preselected amount introduced at said transmitter station by said phase transformation means to obtain a group of inverse phase transformed sum components,

means for generating from said supplementary signal a plurality of harmonics of said frequency component of said supplementary signal, wherein said plurality of harmonics have amplitudes and phase angles respectively equal to said predetermined uniform amplitude and said predetermined uniform phase angle of said frequency components of said pitch signal, and said plurality of harmonics are in one-to-one correspondence with said group of inverse phase transformed sum components, and

means for subtracting each of said plurality of harmonics from a corresponding one of said inverse phase transformed sum components to obtain a plurality of reconstructed frequency components having amplitudes that closely follow the amplitudes of corresponding selected frequency components of said speech wave.

9. Apparatus as defined in claim 8 wherein said means for generating from said supplementary signal a plurality of harmonics of said frequency component of said supplementary signal comprises a low pass filter, a harmonic generator, and a variable delay element connected in series.

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