

Dec. 16, 1969

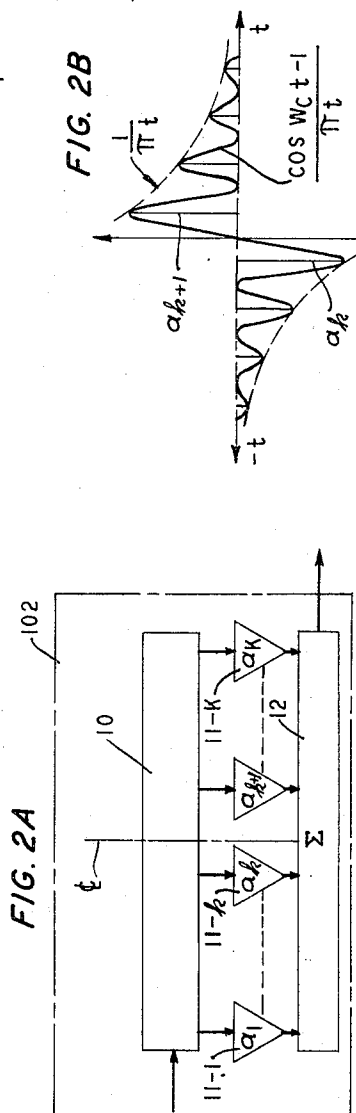
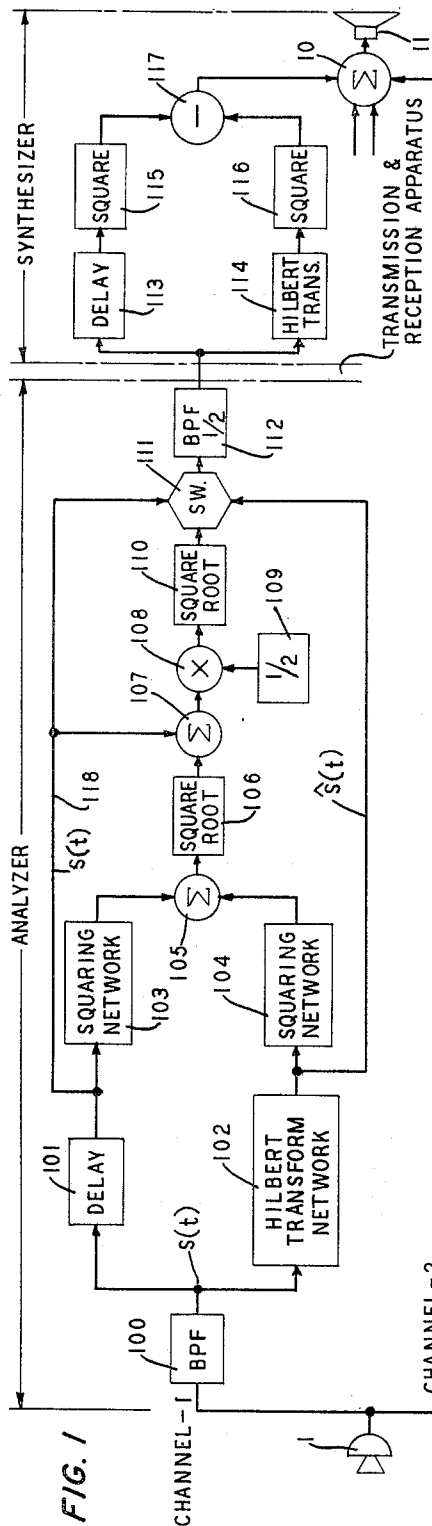
J. L. FLANAGAN ET AL

3,484,556

BANDWIDTH COMPRESSION ELIMINATING FREQUENCY TRANSPOSITION
AND OVERCOMING PHASE AMBIGUITY

Filed Nov. 1, 1966

3 Sheets-Sheet 1



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3 Sheets-Sheet 2

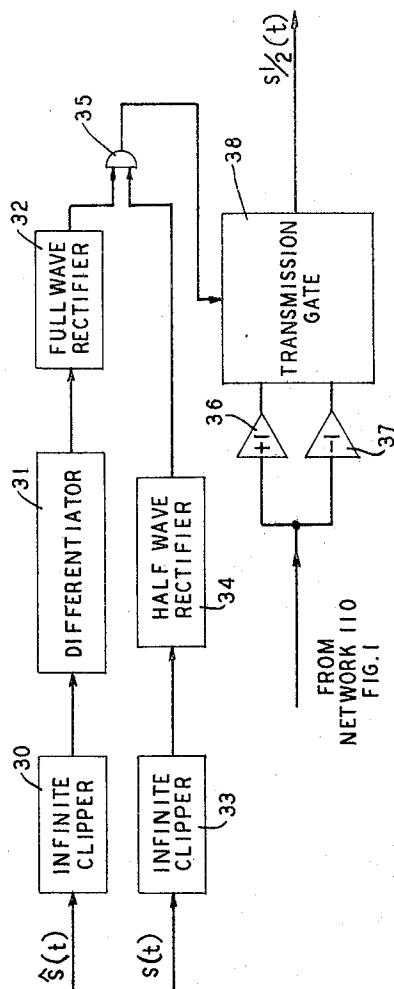


FIG. 3

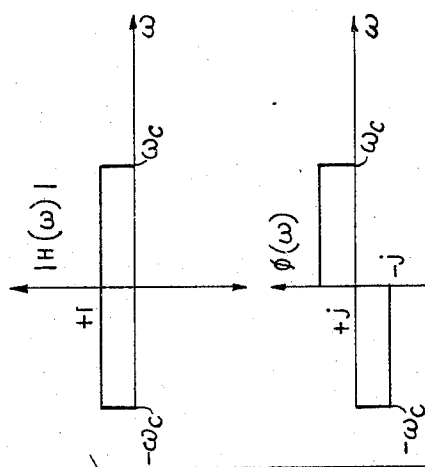
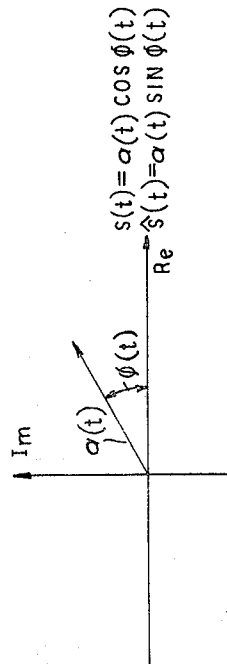


FIG. 2C

FIG. 2D



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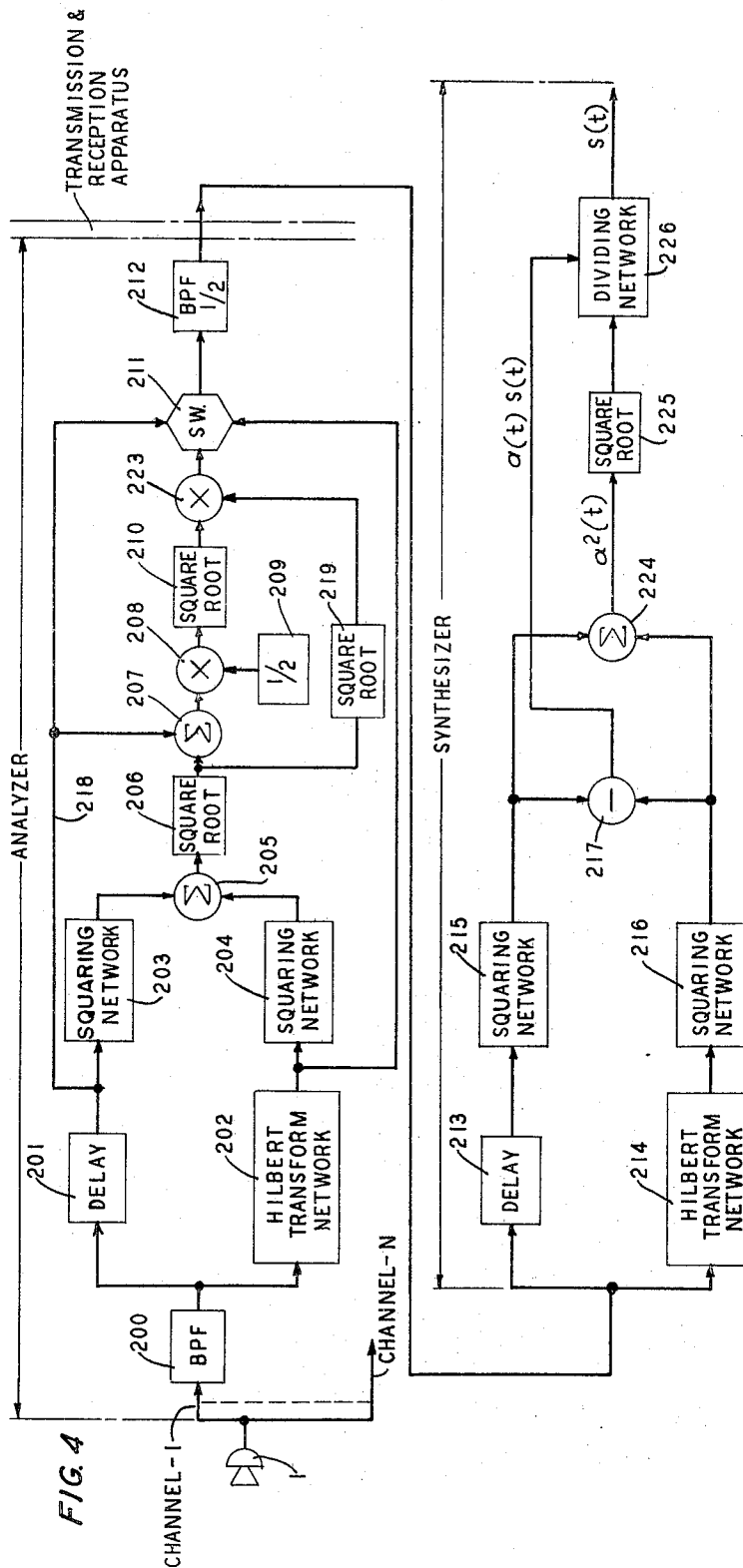
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3 Sheets-Sheet 3



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BANDWIDTH COMPRESSION ELIMINATING FREQUENCY TRANSPOSITION AND OVERCOMING PHASE AMBIGUITY

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10 Claims

ABSTRACT OF THE DISCLOSURE

A system for speech bandwidth compression is disclosed in which the speech is separated into contiguous frequency bands representing the formants of speech. Each band is assumed to be the real part of an analytic signal, and the Hilbert transform taken to represent the imaginary part. From these components the real part of the square root of the analytic signal is synthesized and then filtered to reduce its bandwidth by one-half. Each band-reduced signal is then transmitted by conventional methods. At the receiver each signal is again assumed to be the real part of an analytic signal and the Hilbert transform taken to represent the imaginary part. From these components an approximate replica of each band is reconstructed. All of the bands are summed to reproduce the speech input.

This invention relates to speech bandwidth compression and in particular to the reduction of the bandwidth needed to transmit speech signals by treating the speech signals, or subsignals derived therefrom, as analytic signals.

Prior attempts to compress the bandwidth of speech signals by the technique known as analytic signal processing have been unsuccessful. The synthesized speech contains "burbles," consonants are unintelligible or missing, and transitions from one sound to another are blurred. Various reasons have been advanced for this.

One is that prior art systems, as exemplified by the system disclosed by J. Daguette, in an article entitled "Systeme de Compression de la Parole 'Codimex'," published in Vol. II, Proceedings of the Stockholm Speech Communications Seminar, Royal Institute of Technology, Stockholm, Sweden, distort the output speech because of the manner in which they implement the analytic signal processing. For example, all such known systems transpose the speech signal, or subsignals derived therefrom, to a higher frequency band and carry out the analytic signal processing at these higher frequencies. Unfortunately, in the presence of noise the frequency transposition makes it difficult, if not impossible, to resolve phase ambiguity inherent in the band-compressed signal.

Accordingly, an object of this invention is to eliminate the need for transposing a signal in frequency prior to compressing its bandwidth.

Another object is to accurately resolve any phase ambiguity in the band-compressed signal.

Another reason advanced for the unsuccessful performance of prior art analytic speech compressors is that the received narrowband signals are filtered to remove unwanted frequency components during the synthesis of a replica of the input speech. This filtering introduces additional distortion in the output speech signal.

Accordingly, another object of this invention is to synthesize an accurate replica of the input speech signal from the transmitted narrowband signals without further de-

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grading the quality of the reconstructed speech by filtering during the synthesis process.

These and other related objects are obtained in this invention by dividing a speech signal into several subsignals and then band compressing the analytic versions of each of these subsignals directly, without transposing them to a higher carrier frequency. As a result of eliminating the frequency transposition, the instantaneous phases of the real parts of the band compressed signals can be accurately determined at all times, eliminating one of the sources of degradation in the prior art systems.

Further, according to this invention, a replica of the original speech signal is synthesized by a procedure which requires no filtering and does not, per se, degrade the quality of the resulting speech signal.

In one embodiment of this invention, the input speech signal is divided into a plurality of subsignals occupying contiguous frequency bands. Each subsignal is treated as a separate analytic signal containing both real and imaginary parts and the real part of the square root of each analytic signal is obtained. The uncertainty in algebraic sign of each square rooted signal, inherent in taking the square root, is accounted for by changing the sign of each square root signal every time the signal representing the imaginary part of the analytic signal passes through zero while the real part of the analytic signal is positive. Noise, rather than introducing uncertainty in the number of zero crossings of this signal merely changes slightly the times at which it crosses zero amplitude.

The signal representing the real part of the square root of each analytic signal is filtered prior to transmission, as in prior art systems, to confine this signal to half its original bandwidth. It is then transmitted to the receiver by well-known analog or digital techniques.

At the receiver, a replica of each subsignal is obtained by generating a corresponding quadrature signal from each received narrowband real signal, squaring these two signals, and subtracting the squared quadrature signal from the squared real signal. The sum of the resulting subsignals is a very close replica of the input speech signal. No filtering to remove unwanted frequencies or frequency transposing is necessary in this synthesis.

In another embodiment of this invention, the amplitude of the real part of the square root of each analytic signal is restored to its original value prior to transmission. This provides a method for analytically dividing the instantaneous frequency of the original signal and again makes possible its transmission at reduced bandwidth.

This invention may be more fully understood from the following detailed description in which:

FIG. 1 is a schematic block diagram of a speech compression system constructed according to the principles of this invention;

FIGS. 2A, 2B, 2C, and 2D are of use in explaining the derivation and characteristics of an analytic signal;

FIG. 3 is a schematic block diagram of switch 111 shown in FIG. 1; and

FIG. 4 is a schematic block diagram of an alternative embodiment of this invention.

In this invention an input speech signal is divided into several subsignals occupying contiguous frequency bands. Each subsignal $s(t)$ is treated as the real part of an "analytic signal"

$$\sigma(t) = s(t) + j\hat{s}(t) \quad (1)$$

where $\hat{s}(t)$, the so-called Hilbert transform of $s(t)$, contains the same frequency components as $s(t)$ shifted in phase by $\pi/2$ radians.

Since, as shown in FIG. 2D, $\sigma(t)$ can also be written as

$$\sigma(t) = a(t)e^{j\phi(t)} \quad (2)$$

where $a(t)$, the instantaneous amplitude of $\sigma(t)$, equals $\sqrt{s(t)^2 + \hat{s}(t)^2}$, and $\Phi(t)$, the instantaneous phase of $\sigma(t)$, equals $\tan^{-1}\hat{s}(t)/s(t)$, it is seen that the n^{th} root of the analytic signal $\sigma(t)$ is merely

$$\sigma(t)^{1/n} = [a(t)]^{1/n} e^{j\Phi(t)/n} \quad (3)$$

Thus the analytic signal $\sigma(t)^{1/n}$ representing the n^{th} root of $\sigma(t)$ is seen to have an instantaneous phase $(1/n)^{\text{th}}$ that of $\sigma(t)$. Unfortunately, the bandwidth of $\sigma(t)^{1/n}$ is not in general $(1/n)^{\text{th}}$ that of $\sigma(t)$. Thus $\sigma(t)^{1/n}$ possesses energy outside the desired frequency range. However, for sufficiently small n (say, $n=2, 3, 4$) and most narrow-band speech signals, most of the energy of $\sigma(t)^{1/n}$ is contained in a bandwidth $(1/n)^{\text{th}}$ that of $\sigma(t)$. For such signals, bandwidth compression by a factor of n can be achieved and an approximate replica of $s(t)$ can be constructed by raising the analytic representation of the transmitted signal to the corresponding power n .

This invention is based on the discovery that when $n=2$ a relatively simple apparatus will yield an analytic signal with an instantaneous phase precisely one half the instantaneous phase of $\sigma(t)$. No frequency transposing is required to obtain this band-compressed representation of $\sigma(t)$. Further, it has been discovered that when $n=2$, an approximate replica of $s(t)$ can be synthesized by carrying out some relatively simple operations on the real part of the square root of $\sigma(t)$.

From Equation 3, $\sigma(t)^{1/2}$ can be written as

$$\sigma(t)^{1/2} = \sqrt{a(t)} [\cos \Phi(t)/2 + j \sin \Phi(t)/2] \quad (4)$$

or, equivalently

$$\sigma(t)^{1/2} = s_{1/2}(t) + j\hat{s}_{1/2}(t) \quad (5)$$

where the terms $s_{1/2}(t)$ and $\hat{s}_{1/2}(t)$ are the real and imaginary parts, respectively, of $\sigma(t)^{1/2}$. Thus

$$s_{1/2}(t) = \sqrt{a(t)} \cos \Phi(t)/2 \quad (6)$$

and

$$\hat{s}_{1/2}(t) = \sqrt{a(t)} \sin \Phi(t)/2 \quad (7)$$

The imaginary part $\hat{s}_{1/2}(t)$ of $\sigma(t)^{1/2}$ is just the Hilbert transform of the real part $s_{1/2}(t)$.

Making use of the trigonometric identities

$$\cos \alpha/2 = \pm \sqrt{1/2(1 + \cos \alpha)}$$

and

$$\sin \alpha/2 = \pm \sqrt{1/2(1 - \cos \alpha)}$$

and the definition of $a(t)$ given above, the following equations for the real and imaginary parts of the square root of the analytic signal $\sigma(t)$ are obtained:

$$s_{1/2}(t) = \pm (1/2)^{1/2} [a(t) + s(t)]^{1/2} \quad (8a)$$

and

$$\hat{s}_{1/2}(t) = \pm (1/2)^{1/2} [a(t) - s(t)]^{1/2} \quad (8b)$$

The inverse operation to analytic signal rooting is given by

$$s(t) = \text{Re} [s_{1/2}(t) + j\hat{s}_{1/2}(t)]^2 = \text{Re} [s_{1/2}^2(t) + 2js_{1/2}(t)\hat{s}_{1/2}(t) - \hat{s}_{1/2}^2(t)] \quad (9)$$

where "Re" stands for "real part." Thus a replica $s'(t)$ of the original subsignal $s(t)$ is obtained from the following equation:

$$s'(t) = [s_{1/2}^2(t) - \hat{s}_{1/2}^2(t)] \quad (10)$$

Substituting Equations 8a and 8b into Equation 10 yields the desired relationship that

$$s'(t) = s(t) \quad (11)$$

Thus an approximate version of $s(t)$ can be reconstructed at a synthesizer by merely transmitting to the synthesizer the real part $s_{1/2}(t)$ of the square root of the analytic signal $\sigma(t)$ representing $s(t)$, deriving from this real part at the synthesizer its Hilbert transform $\hat{s}_{1/2}(t)$, and operating on $s_{1/2}(t)$ and $\hat{s}_{1/2}(t)$ as required by Equation 10.

FIG. 1 shows one embodiment of such a band compression system. An input speech signal, converted into an electrical signal by transducer 1, is divided by band-

pass filters 100 into a plurality of N subsignals occupying contiguous frequency bands, where N is a selected positive integer. Each subsignal ideally contains no more than one formant. FIG. 1 shows in detail the apparatus provided to process one such subsignal. Each of the other subsignals is processed by similar apparatus, omitted from FIG. 1 for the sake of clarity.

The subsignal transmitted by filter 100 is sent along two parallel paths. Hilbert transform network 102 is placed in one path while delay 101, equal to one half the maximum delay associated with the Hilbert network 102, is placed in the other path.

Before describing network 102 a word is in order on the theory behind this network. The Hilbert transform $\hat{s}(t)$ of a band-limited signal $s(t)$ merely contains all positive frequencies in $s(t)$ advanced by $\pi/2$ radians and all negative frequencies in $s(t)$ retarded by $\pi/2$ radians. Therefore, the impulse response, within an arbitrary sign, of the band-limited Hilbert transform shown in FIG. 2C is just $(\cos \omega_c t - 1)/\pi t$, where ω_c is the cutoff frequency of $s(t)$. Thus the time response of network 102 (FIG. 1) to an impulse at zero time is shown by the solid line in FIG. 2B, again within an arbitrary sign.

One possible embodiment of Hilbert network 102, shown in FIG. 2A, contains a tapped delay line 10 with an input lead and K output taps uniformly spaced in time, where K is a selected positive integer. According to the Nyquist sampling theorem, a continuous band-limited signal in the time domain can be represented by samples obtained at intervals separated in time by $1/2f$ seconds, where f is the highest significant frequency component in the signal to be sampled. However, because the impulse response of the band-limited Hilbert network is zero at every even Nyquist sampling time 0, $2\pi/\omega_c$, $4\pi/\omega_c$, . . . etc., the output taps on delay line 10 are placed apart by $2\pi/\omega_c$ seconds, at positions corresponding to the odd Nyquist sampling times.

The amplitudes of the signals detected at the output taps of the delay line 10 are set by amplifiers 11-1 through 11- K to be proportional to the amplitudes of the corresponding samples of the impulse response of the Hilbert network 102. These samples are shown as vertical lines in FIG. 2B. The output signals from these amplifiers are summed in summing network 12 to produce a signal $\hat{s}(t)$ representing the Hilbert transform of $s(t)$.

A band-limited signal, such as $(\cos \omega_c t - 1)/\pi t$, theoretically is not truncated in time. Here though, the impulse response of network 102 is terminated after a selected time $2T$ because delay line 10 must be kept to a practical length. T is one half the maximum delay of line 10 and is selected to ensure that the output signal $\hat{s}(t)$ from network 102 is substantially equivalent to the Hilbert transform of $s(t)$.

While the response $(\cos \omega_c t - 1)/\pi t$ of network 102 to an impulse at zero time exists in both negative and positive time relative to zero time, a physically realizable network produces an output signal only in positive time. Thus a signal $s(t)$ must be delayed an amount equal to one half the delay time of delay line 10 to synchronize it with its Hilbert transform $\hat{s}(t)$ from network 102. Delay 101 does this.

From Equation 8a it is seen that the real part $s_{1/2}(t)$ of $\sigma(t)^{1/2}$ is defined in terms of the envelope $a(t)$ of the analytic signal $\sigma(t)$ as well as in terms of $s(t)$. $a(t)$ was defined above as

$$a(t) = \sqrt{s(t)^2 + \hat{s}(t)^2} \quad (12)$$

Therefore, squaring networks 103 and 104 square $s(t)$ and $\hat{s}(t)$, respectively. These networks might, for example, be nonlinear function generating networks of well-known design. The squared signals are combined in summing network 105 and their square root $a(t)$ is obtained from square root network 106, likewise of well-known design.

According to Equation 8a, $a(t)$ must be added to $s(t)$

to obtain $s_{1/2}(t)$, the signal which when filtered occupies approximately one half the bandwidth of $s(t)$. Thus, at the output of delay 101, $s(t)$ is passed on lead 118 to summing network 107 where it is added to $a(t)$. Product network 108 multiplies the resulting sum signal by one half as required by Equation 8a and square root network 110 takes the square root of the resulting product signal.

The resulting signal represents $s_{1/2}(t)$ except for one difficulty; the sign of $s_{1/2}(t)$ is undetermined. This ambiguity is removed by switching network 111.

FIG. 3 shows switching network 111 in more detail. Network 111 detects every zero crossing of $s(t)$ when $s(t)$ is positive. At each of these zero crossings a pulse is generated which changes the sign of the output signal $|s_{1/2}(t)|$ from square root network 110. The output signal from switch 111 thus represents $s_{1/2}(t)$ within an arbitrarily specified initial sign.

In FIG. 3, infinite clipper 30 clips the signal $s(t)$ representing the imaginary part of $\sigma(t)$. The clipped signal is differentiated by differentiator 31 to yield a positive output pulse when the amplitude of $s(t)$ goes from negative to positive and a negative output pulse when the amplitude of $s(t)$ goes from positive to negative. Full-wave rectifier 32 produces a series of positive voltage spikes in response to the alternating positive and negative voltage spikes from differentiator 31. Infinite clipper 33 clips $s(t)$, the real part of the analytic signal $\sigma(t)$. Half-wave rectifier 34 passes the positive portions of the alternating square wave produced by clipper 33. Thus, AND gate 35, with input leads connected to the output leads from rectifiers 32 and 34, produces an output pulse every time $s(t)$ crosses zero amplitude when $s(t)$ is positive.

The signal from AND gate 35 is used to repetitively switch the signal from square root network 110 (FIG. 1) from positive to negative sign and vice versa. Thus, for example, a pulse from AND gate 35 activates transmission gate 38 such that the output signal from positive unity gain amplifier 36 is passed through gate 38 for transmission to the speech synthesizer. On the next pulse from AND gate 35, transmission gate 38 passes the signal from negative unity gain amplifier 37 for transmission to the speech synthesizer. Thus, the sign of $|s_{1/2}(t)|$ is automatically switched every time the analytic signal $\sigma(t)$ increases in phase by 2π radians.

The output signal from switch 111 (FIG. 1) is passed through bandpass filter 112 with one half the bandwidth of bandpass filter 100. Filter 112 removes any undesired high frequency components of $s_{1/2}(t)$ and thus allows a signal representing the real part of the analytic signal $\sigma(t)^{1/2}$ to be transmitted over a bandwidth one half the bandwidth of $s(t)$.

A replica of the input speech signal is obtained at a synthesizer (FIG. 1) consisting of delay 113, Hilbert transform network 114, squaring networks 115 and 116, subtracting network 117, summing network 10, and transducer 11. Since a signal representing only the real part $s_{1/2}(t)$ of the square rooted analytic signal $\sigma(t)^{1/2}$ is transmitted, network 114 is used to obtain the Hilbert transform $\hat{s}_{1/2}(t)$ of this transmitted signal. Network 114 is similar in construction to network 102 shown in FIG. 2A except its cutoff frequency ω_c is one half that of network 102. Delay 113 is equal to one half the maximum delay of the delay line in network 114. The output signal from network 114, $\hat{s}_{1/2}(t)$, is squared in network 116, and $s_{1/2}(t)$ is squared in network 115 after it emerges from delay 113. Subtracting network 117 subtracts $\hat{s}_{1/2}^2(t)$ from $s_{1/2}^2(t)$ to yield, as dictated by Equation 10, the signal $s'(t)$. Because of filter 112, $s'(t)$ is an approximate replica of subsignal $s(t)$.

The process described above is simultaneously carried on by similar apparatus to yield, at the speech synthesizer, replicas of the other subsignals derived at the analyzer from the input speech signal. All such similarly reconstructed subsignals are combined in summing network

10 to produce a replica of the input signal. This replica is converted in transducer 11, for example a loudspeaker, into an acoustic signal representing the input acoustic signal.

While FIG. 1 shows the input speech signal being divided into a plurality of subsignals prior to processing, the input speech signal can, if desired, be processed directly without being divided into subsignals. This has the desirable effect of removing crosstalk between subsignals. But balancing this benefit is the fact that the signal representing the square root of the analytic speech signal has greater energy in higher bandwidths which is lost when its real part $s_{1/2}(t)$ is passed through the equivalent of filter 112 (FIG. 1).

FIG. 4 shows a version of this system which restores $a(t)^{1/2}$, the amplitude of the real part of the square root of the analytic signal $\sigma(t)$, to its original amplitude $a(t)$ prior to transmission. This system makes possible instantaneous frequency division by means of analytic signal processing. This system appears advantageous when the bandpass filters 100 (FIG. 1) have relatively narrow bandwidths such that the signals passed through each bandpass filter vary relatively slowly in amplitude.

The analyzer shown in FIG. 4 is identical to the analyzer shown in FIG. 1 except that, prior to passing the signal $s_{1/2}(t)$ through switch 211, this signal is multiplied by $[a(t)]^{1/2}$ in product network 223. The term $a(t)^{1/2}$ is obtained by taking the square root in network 219 of the output signal from square root network 206.

At the synthesizer it is necessary to compensate for the increase in the amplitude of the transmitted narrowband signal to ensure that the signal $s(t)$ from subtracting network 117 (FIG. 1) is of proper amplitude relative to the amplitudes of the other subsignals.

Thus, at the synthesizer, the received narrowband signal representing $a(t)^{1/2}s_{1/2}(t)$ is transmitted through Hilbert network 214 and simultaneously, through synchronizing delay 213. Since the received narrowband signal can be written as $a(t) \cos \Phi(t)/2$, the output signal from Hilbert network 214 is merely $a(t) \sin \Phi(t)/2$, for the case where $a(t)$ is a slowly varying signal.

The output signals from delay 213 and Hilbert network 214 are squared in network 215 and 216 and then subtracted as shown in subtracting network 217 to yield the signal $a(t)s(t)$. The signals from networks 215 and 216 are also added in summing network 224 to produce an output signal proportional to $a(t)^2$. The square root of this signal is produced in network 225. Then $a(t)s(t)$ is divided by $a(t)$, the output signal from network 225, in dividing network 226 to produce a signal proportional to $s(t)$.

A replica of the input speech is produced by summing all the similarly processed subsignals $s(t)$ as in FIG. 1 and then converting the resulting electrical signal into an acoustic replica of the input speech.

Other embodiments of this invention will be obvious to those skilled in analytic signal processing.

What is claimed is:

1. Apparatus which comprises

means for dividing a speech signal into N subsignals,

where N is a selected positive integer, greater than 1,

means for representing the n^{th} of said subsignals as the real part of an analytic signal, where n is an integer with a value given by $1 \leq n \leq N$,

means for constructing the real part of the square root of said analytic signal,

means for repetitively changing the sign of said real part,

means for filtering said real part,

means for transmitting said filtered real part to a synthesizer,

means at said synthesizer for deriving a replica of said n^{th} subsignal from said transmitted signal, and

means for combining N such similarly produced replica

subsignals to produce a replica of said input speech signal.

2. Apparatus which comprises

means for dividing a speech signal into N subsignals $1 \dots n, \dots N$, where N is a selected positive integer and n is an integer with a value given by $1 \leq n \leq N$,

means for producing a first set of signals which represent the real and imaginary parts of N analytic signals representing said N subsignals,

first means for processing said first set of signals to produce N intermediate signals proportional to the real parts of the square roots of said analytic signals,

means for changing the algebraic sign of the n^{th} intermediate signal each time the signal representing the imaginary part of the n^{th} analytic signal passes through zero amplitude with said n^{th} subsignal positive,

means for filtering said N intermediate signals,

means for transmitting said N filtered intermediate signals to a synthesizer; and
at said synthesizer,

means for generating a second set of signals representing the real and imaginary parts of N analytic signals representing said N filtered intermediate signals,

second means for processing said second set of signals to produce N replicas of said N subsignals, and

means for combining said N replicas to produce a replica of said speech signal.

3. Apparatus as in claim 2 in which said means for changing the algebraic sign of said n^{th} intermediate signal comprises

means for producing a first signal when the signal from said first set, representing the imaginary part of the n^{th} analytic signal, crosses zero amplitude,

means for producing a second signal when the signal from said first set, representing the real part of said n^{th} analytic signal, has a positive amplitude, and

means for changing the algebraic sign of said n^{th} intermediate signal upon simultaneous presence of said first signal and said second signal.

4. Apparatus as in claim 3 in which said means for generating, and said second means for processing, comprise

means for obtaining a third set of signals, each signal in said third set representing the Hilbert transform of a corresponding one of said N intermediate signals,

means for delaying said N intermediate signals to synchronize them with their corresponding Hilbert transform signals in said third set,

means for squaring said N delayed intermediate signals, means for squaring the signals in said third set of signals, and

means for subtracting said squared signals in said third set from their corresponding squared intermediate signals to produce N replicas of said N subsignals.

5. Apparatus which comprises

means for dividing an input speech signal into N subsignals, where N is a selected positive integer,

means for delaying the n^{th} of said subsignals by a selected amount, where n is an integer given by $1 \leq n \leq N$,

means for generating a first signal representing the Hilbert transform of said n^{th} subsignal,

means for squaring both said delayed n^{th} subsignal and said first signal,

means for summing said squared first signal and said squared delayed n^{th} subsignal to produce a sum signal,

means for taking the square root of said sum signal, means for adding said delayed n^{th} subsignal to said

square root of said sum signal to produce a second sum signal,

means for multiplying said second sum signal by one half to produce a third signal,

means for taking the square root of said third signal to produce an intermediate signal,

means for repetitively changing the algebraic sign of said intermediate signal to produce a fourth signal representing the real part of the square root of an analytic signal representing said n^{th} subsignal,

means for filtering said fourth signal,

means for transmitting said filtered fourth signal to a synthesizer; and

at said synthesizer,

means for producing a fifth signal representing the Hilbert transform of said received fourth signal,

means for delaying said received fourth signal to synchronize it with said fifth signal,

means for squaring said delayed fourth signal,

means for squaring said fifth signal,

means for subtracting said squared fifth signal from said squared, delayed fourth signal to produce a replica of said n^{th} subsignal, and

means for combining all the similarly obtained replicas of said N subsignals to produce a replica of said input speech signal.

6. Apparatus which comprises

means for dividing a speech signal into N subsignals $1 \dots n, \dots N$, where n is a selected positive integer and n is an integer with a value given by $1 \leq n \leq N$,

means for producing a first set of signals which represent the real and imaginary parts of N analytic signals representing said N subsignals,

first means for processing said first set of signals to produce N intermediate signals proportional to the real parts of the square roots of said analytic signals,

means for squaring the amplitudes of said N intermediate signals,

means for changing the algebraic sign of the n^{th} intermediate signal each time the signal representing the imaginary part of the n^{th} analytic signal passes through zero amplitude with said n^{th} subsignal positive,

means for filtering said N intermediate signals,

means for transmitting said N filtered intermediate signals to a synthesizer; and

at said synthesizer

means for generating a second set of signals representing the real and imaginary parts of N analytic signals representing said N intermediate signals,

second means for processing said second set of signals to produce N replicas of said N subsignals, and

means for combining said N replicas to produce a replica of said speech signal.

7. Apparatus as in claim 6 in which said means for changing the algebraic sign of said n^{th} intermediate signal comprises

means for producing a first signal when the signal from said first set, representing the imaginary part of the n^{th} analytic signal, crosses zero amplitude,

means for producing a second signal when the signal from said first set, representing the real part of said n^{th} analytic signal, has a positive amplitude, and

means for changing the algebraic sign of said n^{th} intermediate signal upon simultaneous presence of said first signal and said second signal.

8. Apparatus as in claim 7 in which said means for generating, and said second means for processing comprise

means for obtaining a third set of signals, each signal in said third set representing the Hilbert transform

of a corresponding one of said N intermediate signals,
 means for delaying said received N intermediate signals to synchronize them with their corresponding Hilbert transform signals,
 means for squaring said N delayed intermediate signals, 5
 means for squaring the signals in said third set of signals,
 means for subtracting said squared signals in said third set from their corresponding squared intermediate signals to produce a fourth set of signals, 10
 means for adding said squared signals in said third set to their corresponding squared intermediate signals to produce a fifth set of signals,
 means for taking the square roots of the signals in said fifth set to produce a sixth set of signals, and 15
 means for dividing the signals in said fourth set by their corresponding signals in said sixth set to produce N replicas of said N subsignals.
 9. Apparatus which comprises 20
 means for dividing an input speech signal into N subsignals, where N is a selected positive integer,
 means for delaying the n^{th} of said subsignals by a selected amount, where n is an integer given by $1 \leq n \leq N$, 25
 means for generating a first signal representing the Hilbert transform of said n^{th} subsignal,
 means for squaring both said delayed n^{th} subsignal and said first signal,
 means for summing said squared n^{th} subsignal and said squared first signal to produce a sum signal, 30
 means for taking the square root of said sum signal to produce a first intermediate signal,
 means for adding said delayed n^{th} subsignal to said first intermediate signal to produce a second sum signal, 35
 means for multiplying said second sum signal by one half to produce a third signal,
 means for taking the square root of said third signal to produce a second intermediate signal, 40
 means for taking the square root of said first intermediate signal to produce a third intermediate signal,
 means for multiplying said second intermediate signal by said third intermediate signal to produce a fourth intermediate signal, 45
 means for repetitively changing the algebraic sign of said fourth intermediate signal to produce a fourth signal representing the real part of the square root of an analytic signal representing said n^{th} subsignal, 50
 means for filtering said fourth signal,
 means for transmitting said filtered fourth signal to a synthesizer; and
 at said synthesizer,

means for producing a fifth signal representing the Hilbert transform of said received fourth signal,
 means for delaying said received fourth signal to synchronize it with said fifth signal,
 means for squaring said delayed fourth signal,
 means for squaring said fifth signal,
 means for subtracting said squared fifth signal from said squared, delayed, fourth signal to produce a sixth signal,
 means for adding said squared fifth signal to said squared, delayed fourth signal to produce a seventh signal,
 means for taking the square root of said seventh signal to produce an eighth signal,
 means for dividing said sixth signal by said eighth and signal to produce a replica of said n^{th} subsignal,
 means for combining all the similarly produced replicas of said N subsignals to produce a replica of said input speech signal.
 10. Apparatus which comprises
 means for converting an acoustic speech signal into an electrical signal,
 means for representing said electrical signal as the real part of an analytic signal,
 means for constructing the real part of the square root of said analytic signal,
 means for repetitively changing the sign of said real part,
 means for filtering said real part,
 means for transmitting said filtered real part to a synthesizer,
 means at said synthesizer for deriving a replica of said electrical signal from said transmitted signal, and
 means for converting said replica signal into a replica of said acoustic speech signal.

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