June 22, 1965 W. LAWRENCE 3,190,960

SPEECH BANDWIDTH COMPRESSION SYSTEMS

Filed Dec. 26, 1961 3 Sheets-Sheet 1

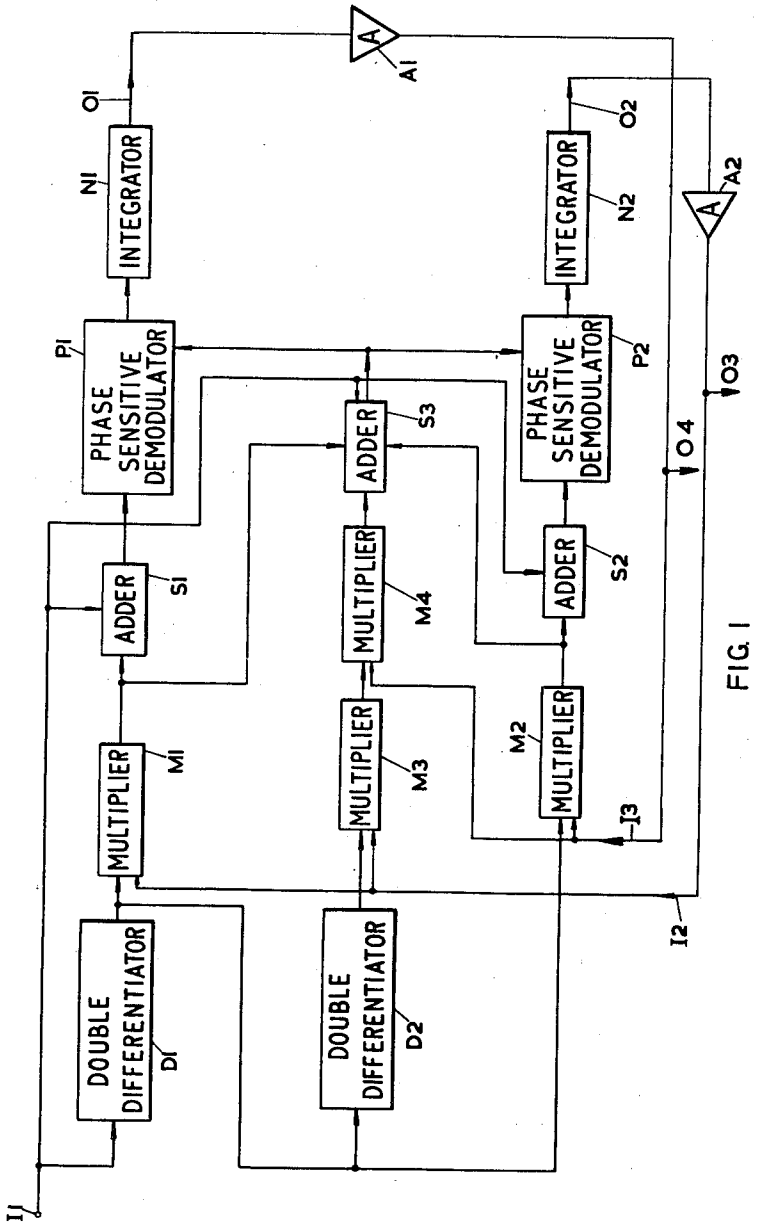


FIG. 1

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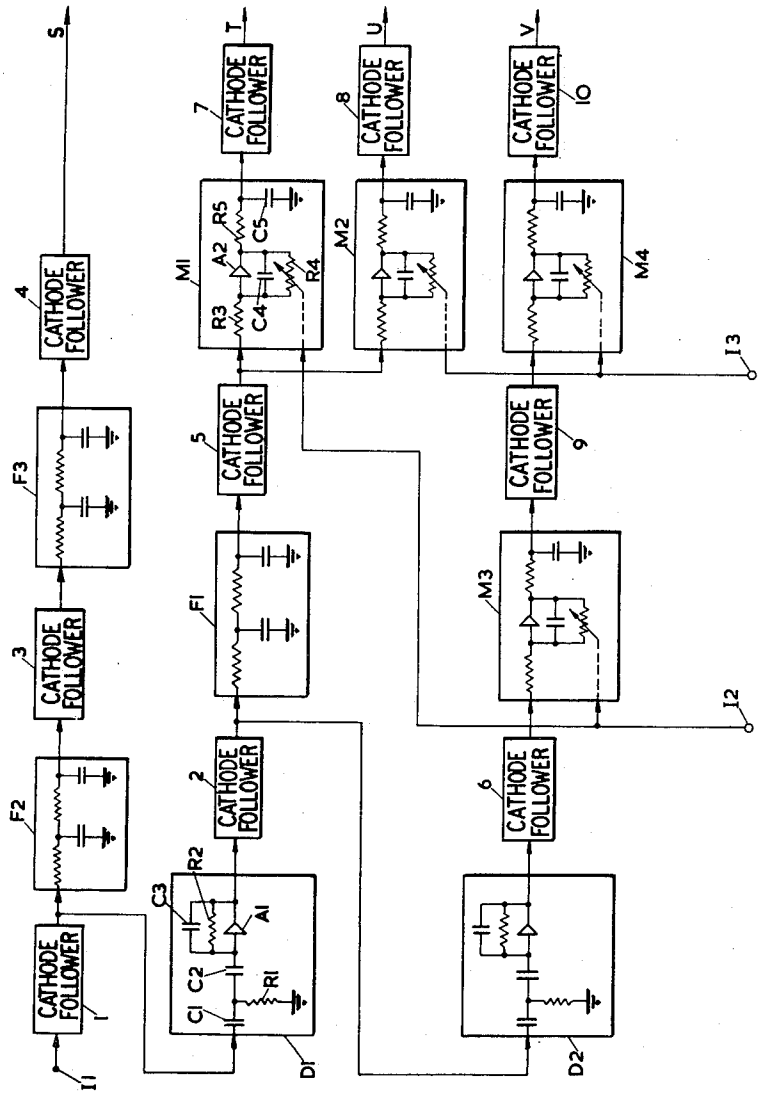


FIG. 2.

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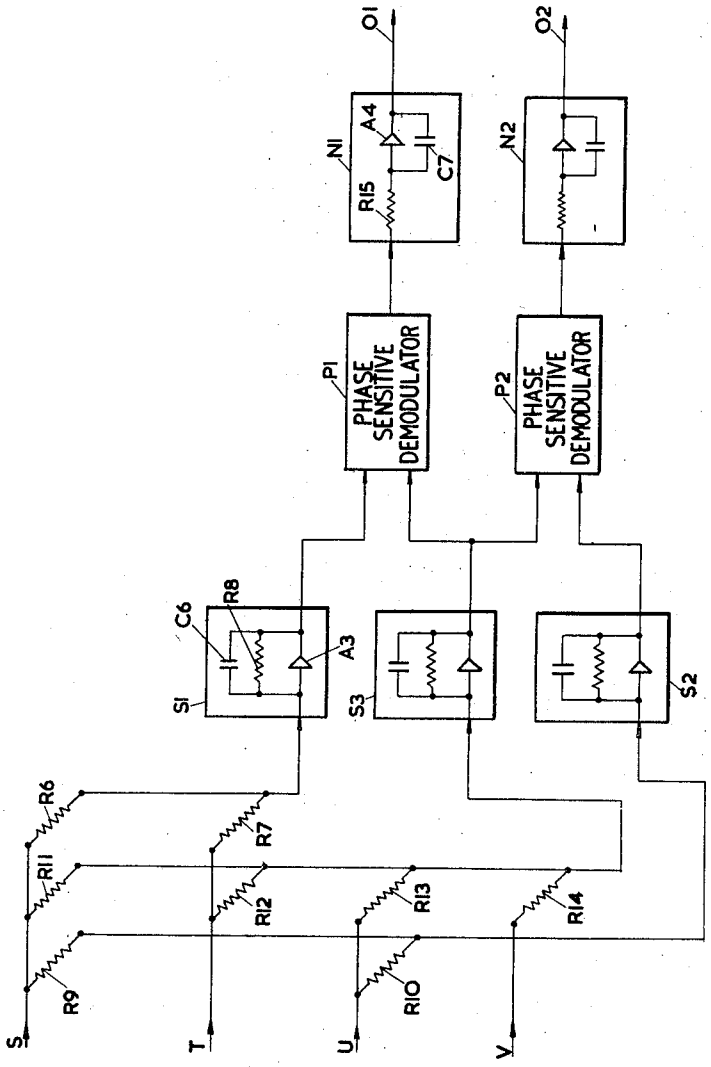


FIG. 2. (cont.)

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SPEECH BANDWIDTH COMPRESSION SYSTEMS

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Claims priority, application Great Britain, Dec. 30, 1960, 44,730/60
7 Claims. (Cl. 179—1)

The present invention relates to speech analysing systems. Such analysing systems may be used in speech bandwidth compression apparatus for the transmission of speech over radio links and the like.

The present invention is based on a theory of speech formation in which it is postulated that the sounds produced in speech are synthesised basically from the production by the vocal chords of a fundamental pitch frequency. This pitch frequency is produced by the opening and closing of the vocal chords, each such opening resulting in a pitch pulse. The pitch pulses emitted from the vocal chords pass through the vocal tract and are subjected there to the action of formant-forming filters in the form of resonators, the resonant frequencies of which are varied as the sounds are articulated. The action of these formant resonators on the pitch pulses may be simulated by passing pitch pulses through a series of resonators or damped oscillators, the oscillation frequencies of which may be varied to produce the required speech. The variable frequencies at which these resonators, or damped oscillators, are operated are called the formant frequencies. In practice, a reasonable likeness to speech may be synthesised by the action on pitch pulses of inter alia a number of formant resonators, or damped oscillators, in series.

An object of the present invention is to produce a formant tracker capable of extracting from speech, formant parameter signals indicative of formant frequencies present in the speech.

It is a further object of the present invention to provide a formant tracker which includes an adjustable network for operating on an input speech signal and tracking means for extracting from a primary output of the network rectified derivatives of the speech signal and for feeding the rectified derivatives back to adjust the network continuously so that the primary output is the speech signal operated upon by substantially the inverse of at least some of the formant-forming filters of the vocal tract, whereby the rectified derivatives are constrained to be representative of the corresponding formant parameters of the input speech signal.

In this specification, the term "formant parameter" is used in the general sense of a parameter which is related in some known manner to a corresponding formant frequency.

A still further object of the invention is to provide a formant tracker for tracking the frequencies of formants formed by formant-forming resonators in a vocal tract during the production of an input speech signal, including an adjustable network means for simulating the inverse of a number n of the said formant-forming resonators over discrete ranges of frequency and having a primary output representative of the said input speech signal operated upon by the electrical analogue of substantially the inverse of the said n formant-forming resonators and having n secondary outputs, each secondary output being representative of the said input speech signal operated upon by the electrical analogue of substantially the inverse of a different number $(n-1)$ of the said n formant-forming resonators, n phase-sensitive demodulators connected in parallel to the said primary output and each connected to a different one of the said n secondary outputs, and means for feeding back output signals from the said phase-sensitive demodulators to adjust the electrical network so that the said output signals are representative of the formant frequencies of the said n formant-forming resonators.

An embodiment of the present invention will now be described, by way of example, with reference to the accompanying drawings in which:

FIGURE 1 is a block schematic diagram of a formant tracker illustrating the basic principles upon which the embodiment of the invention is founded and

FIGURE 2 is a more detailed circuit diagram of the formant tracker shown in FIGURE 1.

Before the drawings are referred to, it must first be explained that the theory underlying the embodiment is based on the assumption that speech is formed by the action on each pitch pulse of a number of serial operations. For simplicity in explaining the invention, only the action of formant-forming filters or resonators on the pitch pulses will be hereinafter considered in detail. The production of the speech signal may be expressed approximately by the operational equation:

$$S(t)=F(t).N(p).H(p) \quad (1)$$

where p is the Heaviside operator, a derivative operator representing d/dt. Where $S(t)$ is a function representing the final spech signal, $F(t)$ is a function representing the pitch pulse, $H(p)$ is an operational function representing the action of all the formant resonators in series and $N(p)$ is an operational function of all the other factors operating in the vocal tract.

It has been found that a good approximation to the action of the formant resonators may be expressed in the form:

$$H(p)=\Pi_1{}^r(h_1(p)).\Pi_r{}^n(h_1(p)) \quad (2)$$

where $h_1(p)=1/(1+p^2/\omega_1{}^2)$, $h_1(p)$ being a transfer function representing the action of one formant resonator at a resonant angular frequency ω_1 which varies with time.

It is found that a reasonable approximation to the function $H(p)$ may be obtained by taking the first two lower frequency formants represented in the series $h_1(p)$. In that case,

$$H(p)=[1/(1+p^2/\omega_1{}^2)][1/(1+p^2/\omega_2{}^2].\Pi_3{}^n(h_i(p)) \quad (3)$$

where ω_1 and ω_2 represent the variable frequencies of the first two formants.

Clearly, if the first three formants are taken, then

$$H(p)=[1/(1+p^2/\omega_1{}^2][1/(1+p^2/\omega_2{}^2)][1/(1+p^2/\omega_3{}^2)].\Pi_4{}^n(h_i(p)) \quad (4)$$

and so on if more formants are taken.

From Equations 1 and 3:

$$S(t)=P_1(t).[1/(1+p^2/\omega_1{}^2)][1/(1+p^2/\omega_2{}^2)] \quad (5)$$

where $P_1(t)=F(t).N(p).\Pi_3{}^r(h_i(p))$.

Similarly, if three formants are taken, then from Equations 1 and 4

$$S(t)=P_2(t)[1/(1+p^2/\omega_1{}^2)][1/(1+p^2/\omega_2{}^2)][1/(1+p^2/\omega_3{}^2)] \quad (6)$$

where $P_2(t)=F(t).N(p).\Pi_4{}^r(h_i(p))$, and so on if more formants are taken.

In the following description, the function $S(t)$ is used to denote the electrical equivalent of the acoustic signal.

The embodiment about to be described with reference to FIGURES 1 and 2 is a formant tracker which extracts from an electrical signal $S(t)$ two formant parameter signals which are related to the two formant angular frequencies ω_1 and ω_2 of Equation 5.

FIGURE 1 is a block schematic diagram illustrating the principles on which a circuit network for tracking two formant parameters ω_1 and ω_2 is based. FIGURE 1

shows an input terminal I1 to which is applied a speech signal $S(t)$. This speech signal is passed from the input terminal to a double differentiator D1 in which the signal $S(t)$ is subjected to a double differentiation process. By this means the signal $S(t)$ is reversed in phase and given an amplitude weighting which is proportional to the square of the frequency. That is to say that in the double differentiator, the speech signal $S(t)$ is subjected to a transfer function proportional to p^2 so that the output of the double differentiator D1 is a signal proportional to $p^2.S(t)$.

The output of the double differentiator D1 is applied to one input of a multiplier M1 in which it is multiplied by a voltage a applied to the other input of the multiplier from an input terminal I2. It follows that the output of the multiplier M1 is a signal proportional to $a.p^2S(t)$. This is added in an adder S1 to the original signal so that the output of the adder S1 is a signal proportional to $(1+a.p^2).S(t)$. The output of the double differentiator D1 is also applied to one input of a multiplier M2 to the other input of which a voltage b is applied via an input terminal I3. The output of the multiplier M2 is a signal proportional to $b.p^2.S(t)$. This is added in an adder S2 to the original signal so that the output of the adder S2 is a signal proportional to $(1+b.p^2).S(t)$.

The output of the double differentiator D1 is also applied to a second double differentiator D2, similar to the double differentiator D1, so that the output of the double differentiator D2 is a signal proportional to $p^4.S(t)$. This signal is applied to one input of a multiplier M3, to the other input of which is applied the voltage a via the terminal I2. The output of the multiplier M3 is, therefore, a signal proportional to $a.p^4.S(t)$. This signal is applied to one input of a further multiplier M4 to the other input of which is applied the voltage b via the input I3. The output of the multiplier M4 is, therefore, a signal proportional to $a.b.p^4.S(t)$. The outputs of the multipliers M1, M2 and M4 and the original signal are all applied to a four-input adder S3 so that the output of this adder is proportional to

$$[1+(a+b)p^2+a.b.p^4]S(t) = (1+ap^2)(1+bp^2).S(t) \quad (7)$$

It will be seen from Equation 5, that if the assumptions made therein are correct, and a in Expression (7) is made equal to $1/\omega_1^2$ and b is made equal to $1/\omega_2^2$ then the speech signal $S(t)$ will have been operated upon by the inverse of the formant-forming filters in the vocal tract which have resonant frequencies of ω_1 and ω_2 respectively and the output of the adder S3 (referred to as the primary output of the network) will be a signal represented by the function $P_1(t)$ of Equation 5 above. The correct values for the voltages a and b are obtained by the following means. The outputs of the adders S1 (referred to as a secondary output of the network) and S3 are applied to separate inputs of a phase-sensitive demodulator P1, the output of which is applied to an integrator amplifier N1. The output voltage at an output terminal O1 of the integrator amplifier N1 is applied to the input terminal I3 and, as will be explained hereinafter, is then a voltage which is a function of ω_2. Similarly, the outputs of the adders S2 (also referred to as a secondary output of the network) and S3 are applied to separate inputs of a phase-sensitive demodulator P2, the output of which is applied to an integrator amplifier N2. The output voltage at an output terminal O2 of the integrator amplifier N2 is applied to the input terminal I2 and is then a voltage which is a function of ω_1.

The function performed by the phase-sensitive demodulator P1 will now be explained. In the production of the speech signal $S(t)$ each formant resonator was excited into damped oscillation on the occurrence of a pitch pulse $F(t)$. Thus, the resultant speech signal $S(t)$ consists of the original pitch pulse overlaid with, inter alia, damped oscillations at the formant frequencies, the main two of

which are now being considered. The output of the adder S1 is a signal proportional to $(1+ap^2)S(t)$. Now the component $a.p^2.S(t)$ of this signal is in antiphase with the component $S(t)$ and has an amplitude which is proportional to a and to the inverse of the square of the angular frequency ω_1. It follows that at some value of $a(=1/\omega_1^2)$, the component of the signal $S(t)$ due to one formant frequency, f_1 say, will be eliminated from the output of the adder S1. This value of a will have no inverse linear relationship to the square of the formant frequency. Also, when this value of a is obtained, the only components remaining in the output signals of the adder S1 are those due to the other formant frequency, f_2 say, and the function $P_1(t)$. Under the same conditions, the output of the adder S3 will contain a component at the formant frequency f_2. This component at the formant frequency f_2 will be in phase or antiphase with the corresponding component at the output of the adder S1 according to whether b is less than or greater than $1/\omega_2^2$ respectively. Furthermore, it will have an amplitude which will increase as $b-1/\omega_2^2$ increases which will be zero when $b=1/\omega_2^2$. In the phase-sensitive demodulator, the components at the formant frequency f_2 are multiplied together. It follows that, the voltage at the output terminal O1 of the integrator amplifier N1 will vary about some datum value according to whether b is greater than or less than $1/\omega_2^2$.

Similarly, the voltage at the output terminal O2 of the integrator amplifier N2 will vary about some datum value according to whether a is greater than or less than $1/\omega_1^2$. If now, the output terminal O1 is connected to the input terminal I3 and the output terminal O2 is connected to the input terminal I2 through amplifiers A1 and A2 respectively (as shown in FIGURE 1), then the system will act to constrain b to be equal to $1/\omega_2^2$ and a to be equal to $1\omega_1^2$. The values of the voltages a and b, as presented at outputs O3 and O4 respectively, will then be representative of the required two formant parameters.

The block schematic circuit network described with reference to FIGURE 1 is, unfortunately, difficult to realise in practice because pure differentiators cannot be simulated by circuits. Consequently circuits have to be used which have transfer functions similar to the function of differentiation and then due allowance made for their imperfections. Thus, for example, a practical differentiating circuit does not provide an output in which the amplitude weighting (that is to say, the amplification factor) is proportional to frequency no matter how high the frequency, but has an upper frequency limit, or cut-off frequency, above which the amplification factor increases less with frequency and eventually decreases with further increases of frequency. Signal paths containing more than one differentiation circuit will, therefore, suffer from a greater effective attenuation of the signal at the higher frequencies than signal paths containing fewer differentiating circuits. This effect may be offset to some extent by introducing attenuation into those signal paths having fewer than the maximum number of differentiating circuits in any one signal path, to simulate the attenuation introduced by the differentiating circuits in the signal path containing the maximum number of those circuits. The circuit about to be described with reference to FIGURE 2 incorporates circuits for offsetting the effect of the attenuation at higher frequencies introduced by the differentiating circuits.

FIGURE 2 shows an input terminal I1 through which a speech signal $S(t)$ is applied to a cathode follower 1. The output of the cathode follower 1 is applied to a double differentiator D1. This double differentiator consists of a first differentiator C1, R1 and a second differentiator comprising essentially a capacitor C2, a high gain phase-inverting amplifier A1 and a feedback resistor R2. The feedback resistor R2 is shunted by a small-valued capacitor C3 which accentuates slightly the fall off in amplification discussed above. As hereinbefore

stated, the amplifier A1 is a phase-inverting amplifier (that is to say, its output is in anti-phase to its input) so that the output of the double differentiator D1 is in anti-phase to that which one would expect from an ideal double differentiator. However, this is compensated for in the remainder of the circuit.

The transfer function of the double differentiator D1 may be written as $-(p/\omega_{01})^2/(1+p/\omega_c)^2$ where ω_{01} is the angular frequency, corresponding to say 1 kc./s., at which the amplification is unity and ω_c is the angular frequency, corresponding to say 3 kc./s., at which the output is attenuated by 3 db in power.

The output from the double differentiator D1 is applied through a cathode follower 2 to a second double differentiator D2 having the same transfer function as that of the double differentiator D1. It follows that the total transfer function involved in this signal path is

$$(p/\omega_{01})^4/(1+p/\omega_c)^4$$

The output of the double differentiator D1 is also passed, via the cathode follower 2, to a resistance/capacity filter circuit F1 having a transfer function of

$$1/(1+p/\omega_c)^2$$

This filter circuit is designated to have a substantially level frequency response at the lower frequencies but to attenuate the signal progressively at higher frequencies, the attenuation being of the order of 3 db in power at a frequency of 3 kc./s. The transfer function of the double differentiator D1 and the filter F1 together is, therefore,

$$-(p/\omega_{01})^2/(1+p/\omega_c)^4$$

The output of the cathode follower 1 is applied to a filter F2, a cathode follower 3 and a filter F3 in series, each of these filters being identical to the filters F1, so that the transfer function of these two filters F1 and F2 together is $1/(1+p/\omega_c)^4$. This transfer function will be referred to hereinafter by the symbol A. The outputs of the filters F3 and F1 and the double differentiator D2 are applied to cathode followers 4, 5 and 6 respectively. From the foregoing, it will be seen that the outputs of the cathode followers 4, 5 and 6 will be proportional to $A.S(t)$, $-(A.p^2/\omega_{01}^2)S(t)$ and $(A.p^4/\omega_{01}^4)S(t)$ respectively.

The cathode follower 5 is connected at its output to a multiplier M1 the other input to which is the input terminal 12 to which a voltage a is applied. The multiplier M1 is shown in this drawing as being a phase-inverting amplifier to which the output of the cathode follower 5 is applied through a resistor R3. The amplifier A2 is shown as having a feedback resistor R4 which is variable effectively to vary the gain of the amplifier under control of the voltage a applied to the terminal I2. This is a purely symbolic representation of the working of the multiplier which may be a pulsed attenuator multiplier of a type similar to that described in the book "Electronic Analogue Computers," 2nd edition, by Korn and Korn (published by the McGraw Hill Book Company) as pages 269 and 270. However, preferably in this case the pulsing is done in the feedback path of the amplifier A2 so that the resistance R4 is effectively varied by the duty cycle of the pulsing which is controlled by the voltage a applied to the input I2. In order to reduce ripple on the output of the multiplier M1 due to the chopping frequency, a capacitor C4 shunts the resistor R4 and the amplifier A2 is followed by a ripple filter R5, C5. The output of the ripple filter is applied to a cathode follower 7. The output of the cathode follower 7 is a signal substantially proportional to $(a.A.p^2/\omega_{01}^2).S(t)$. The minus sign has now disappeared because the amplifier A2 is a phase-inverting amplifier.

The output of the cathode follower 5 is also connected to one input of a multiplier M2 having a similar construction to the multiplier M1. The other input to the multiplier M2 is the input terminal I3 to which is applied a voltage b. The output of the multiplier M2 is applied to a cathode follower 8 at the output of which appears a signal substantially proportional to $(b.A.p^2/\omega_{01}^2).S(t)$.

A multiplier M3, a cathode follower 9, a multiplier M4 and a cathode follower 10 are connected in series to the output of the cathode follower 6. The multipliers M3 and M4 are similar in construction to the multiplier M1 and have their second inputs connected to the input terminals I2 and I3 respectively. The signal at the output of the cathode follower 10 is, therefore, substantially proportional to $(a.b.A.p^4/\omega_{01}^4).S(t)$.

The outputs (S and T of FIGURE 2) of the cathode followers 4 and 7 are connected to the input of an analogue adder S1 (in the form of a summing amplifier) through equal resistors R6 and R7 respectively. The adder S1 comprises a phase-inverting amplifier A3 having a feedback resistor R8. The resistor R8 is shunted by a capacitor C6 to reduce unwanted higher frequency components which might interfere with the action of a phase-sensitive demodulator P1 which receives the output of the adder S1. The output of the adder S1 is a secondary output signal proportional to $-(1+a.p^2/\omega_{01}^2)A.S(t)$ which is applied to one input of the phase-sensitive demodulator P1.

An adder S2, similar to the adder S1 has the outputs (S and U of FIGURE 2) of the cathode followers 4 and 8 connected to its input through equal resistors R9 and R10 respectively. The output of this adder is, therefore, a secondary output signal proportional to

$$-(1+b.p^2/\omega_{01}^2)A.S(t)$$

which is applied to one input of a phase-sensitive demodulator P2.

The outputs (S, T, U, V of FIGURE 2) of the cathode followers 4, 7, 8 and 10 are connected to the input of an adder S3 through equal resistors R11, R12, R13 and R14 respectively. The adder S3 is of similar construction to the adder S1. The output of the adder S3 is, therefore, a primary output signal proportional to

$$-[A+(a+b)Ap^2/\omega_{01}^2+abA.p^4/\omega_{01}^4].S(t)$$

which is equal to $-[(1+ap^2/\omega_{01}^2)(1+b.p^2/\omega_{01}^2)]A.S(t)$. This signal is applied to the other inputs of the phase-sensitive demodulators P1 and P2. The signals applied to the phase-sensitive demodulators P1 and P2 are, except for the changes in sign and the upper frequency limits imposed by the imperfections of the circuits, the same as those applied to the phase-sensitive demodulators P1 and P2 of FIGURE 1. The output of the phase-sensitive demodulator P1 is applied to the input of an integrator N1. This integrator consists of a phase-inverting amplifier A4 having a feedback capacitor C7, and a resistor R15 through which the integrator input is applied to the amplifier A4. The output of the integrator N1 is applied through its output terminal O1 and an amplifier (not shown) to the input terminal I3 so as to control the formant parameter voltage b. The voltage at the output terminal O1 will then be proportional to the required formant parameter ω_2.

A further integrator N2, similar in construction to the integrator N1, receives the output of the phase-sensitive modulator P2. The output terminal O2 of the integrator N2 is fed back through an amplifier (not shown) to the input terminal I2 to control the formant parameter voltage a. The voltage at the output terminal O2 will then be proportional to the required formant parameter ω_1.

When the apparatus is initially set up to carry out formant tracking, preset controls (not shown) preferably in the multipliers M1, M2, M3 and M4 and the phase-sensitive detectors P1 and P2 must be set up to ensure that the required formants are tracked by the apparatus. That is to say, the apparatus is so preset that the values of the formant parameter voltages a and b lie in their expected ranges.

It will be realised by those versed in the art that the invention is not limited to the specific apparatus described

with reference to FIGURE 1 or FIGURE 2 for operating upon the input speech signal $S(t)$ with the various transfer functions. Thus, for example, the signal proportional to $(1+ap^2)(1+bp^2)S(t)$ obtained at the primary output of the adder S3 may be produced instead by operating on the output signal proportional to $(1+ap^2)S(t)$ at the output of the adder S1 by apparatus, such as a double differentiator and a multiplier in series, having a transfer function bp^2 and adding the resultant signal, in an adder, to the output of the adder S1.

Although the embodiment hereinbefore described with reference to FIGURES 1 and 2 is constructed to track two formant frequencies, clearly the invention is not limited to the tracking of only two formant frequencies but may be applied to the tracking of any number of formant parameters. Any number, *n*, of formant frequencies may be tracked by operating on an input speech signal $S(t)$ by an adjustable network having a primary output which is the input signal operated upon by a transfer function which is the inverse of all of the *n* formant-forming filters of the vocal tract and *n* secondary outputs each of which is the input signal operated upon by a transfer function which is the inverse of all except one (different in each case) of the *n* formant forming filters. Thus for example, by similar means to those already described with reference to FIGURES 1 and 2, functions similar to

$$(1+ap^2)(1+bp^2).S(t)$$
$$(1+ap^2)(1+cp^2)S(t)$$
$$(1+bp^2)(1+cp^2)S(t)$$

and $(1+ap^2)(1+bp^2)(1+cp^2)S(t)$ could be generated where *a*, *b* and *c* are the values of three formant parameter control voltages. The values of these control voltages for controlling the various multipliers employed could then be determined from three phase-sensitive demodulators each fed with the last of these functions and a separate one of the first three of these functions. For example, the value of the parameter control voltage *c* would be determined by the output of a phase-sensitive demodulator fed with signals proportional to the functions

$$(1+ap^2)(1+bp^2).S(t)$$

and $(1+ap^2)(1+bp^2)(1+cp^2).S(t)$.

I claim:

1. A formant tracker including:
(a) a first double differentiator,
(b) a first multiplier connected to an output of the first double differentiator for multiplying the output signal of the first double differentiator by a first electrical signal,
(c) a first adder having inputs connected to the input of the first double differentiator and to the output of the first multiplier,
(d) a second multiplier connected to an output of the first double differentiator for multiplying the output signal of the first double differentiator by a second electrical signal,
(e) a second adder having inputs connected to the input of the first double differentiator and to the output of the second multiplier,
(f) a second double differentiator having its input connected to the output of the first double differentiator,
(g) a third multiplier connected to an output of the second double differentiator for multiplying the output signal of the second double differentiator by the said first electrical signal,
(h) a fourth multiplier connected to an output of the third multiplier for multiplying the output signal of the second multiplier by the said second electrical signal,
(i) a third adder connected to the input of the first double differentiator and to the outputs of the said first, second and fourth multipliers,
(j) a first phase-sensitive demodulator connected to the outputs of the first and third adders,
(k) a first integrator connected to the output of the first phase-sensitive demodulator,
(l) a first amplifier connected to the output of the first integrator to produce the said second electrical signal,
(m) a second phase-sensitive demodulator connected to the outputs of the second and third adders,
(n) a second integrator connected to the output of the second phase-sensitive demodulator, and
(o) a second amplifier connected to the output of the second integrator to produce the said first electrical signal.

2. A formant tracker for tracking the frequencies of formants formed by formant-forming resonators in a vocal tract during the production of an input speech signal, including an adjustable electrical network means having a primary output, means for extracting from the said primary output formant-parameter signals derived from the said input speech signal, and means for feeding back the said formant-parameter signals to adjust the said electrical network means continuously so that a signal at the primary output is the speech signal operated upon by the electrical analogue of substantially the inverse of at least some of the said formant-forming resonators.

3. A formant tracker for tracking the frequencies of formants formed by formant-forming resonators in a vocal tract during the production of an input speech signal, including an adjustable electrical network means for simulating the inverse of at least some of the said formant-forming resonators over discrete ranges of frequencies, the said electrical network means having a primary output, means for extracting from the said primary output formant-parameter signals derived from the input speech signals, and means for feeding back the said formant-parameter signals to adjust the said electrical network means continuously so that a signal at the primary output is the said input speech signal operated upon by the electrical analogue of substantially the inverse of at least some of the said formant-forming resonators.

4. A formant tracker for tracking the frequencies of formants formed by formant-forming resonators in a vocal tract during the production of an input speech signal, including an adjustable electrical network means for simulating the inverse of at least some of the said formant-forming resonators over discrete ranges of frequency, the said electrical network means having a primary output representative of the said input speech signal operated upon by the electrical analogue of substantially the inverse of the said at least some of the formant-forming resonators and having secondary outputs equal in number to the said at least some of the formant-forming resonators, each secondary output being representative of the input speech signal operated upon by the electrical analogue of the inverse of all except a corresponding one of the said at least some formant-forming resonators, a plurality of phase-sensitive demodulators all connected to the said primary output and each connected to a separate one of the said secondary outputs, and means for feeding back output signals from the phase-sensitive demodulators to adjust the electrical network means so that the said output signals are representative of the formant frequencies of the said at least some formant-forming resonators.

5. A formant tracker as claimed in claim 4 and wherein the said adjustable network includes a first operational means connected to an input of the said adjustable network for converting an input signal $S(t)$ to substantially the form $(1+ap^2).S(t)$ at a secondary output thereof, a second operational means connected to the said input for converting the said input signal to substantially the form $(1+bp^2).S(t)$ at a secondary output thereof, where *a* and *b* are variable electrical quantities and *p* is the Heaviside operator, and third operational means for providing a primary output from the said adjustable network which is of substantially the form $(1+ap^2)(1+bp^2).S(t)$.

6. A formant tracker as claimed in claim 5 including

a first phase-sensitive demodulator, means for applying the said secondary output of the said first operational means and the said primary output to the first phase-sensitive demodulator, means for feeding back the output of the first phase-sensitive demodulator to the said adjustable network to control the electrical quantity *b*, a second phase-sensitive demodulator, means for applying the said secondary output of the said second operational means and the said primary output to the second phase-sensitive demodulator and means for feeding back the output of the second phase-sensitive demodulator to the said adjustable network to control the electrical quantity *a*.

7. A formant tracker for tracking the frequencies of formants formed by formant-forming resonators in a vocal tract during the production of an input speech signal, including an adjustable electrical network means for simulating the inverse of a number *n* of the said formant-forming resonators over discrete ranges of frequency, the said electrical network means having a primary output representative of the said input speech signal operated upon the electrical analogue of substantially the inverse of the said *n* formant-forming resonators and having *n* secondary outputs, each secondary output being representative of the said input speech signal operated upon by the electrical analogue of substantially the inverse of (*n*—1) formant-forming resonators, *n* phase-sensitive demodulators connected in parallel to the said primary output and each connected to a different one of the said *n* secondary outputs, and means for feeding back output signals from the said phase-sensitive demodulators for adjusting the electrical network means so that the said output signals are representative of the formant-frequencies of the said *n* formant-forming resonators.

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