United States Patent [19]

Wolf

[11] **3,833,767**

[45] **Sept. 3, 1974**

[54] **SPEECH COMPRESSION SYSTEM**

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[22] Filed: **Dec. 8, 1972**

[21] Appl. No.: **313,239**

Related U.S. Application Data

[63] Continuation-in-part of Ser. No. 191,003, Oct. 20, 1971, which is a continuation-in-part of Ser. No. 155,988, June 23, 1971.

[52] **U.S. Cl.** ..... **179/15.55 R,** 179/15 BC

[51] **Int. Cl.** **H04b 1/66**

[58] **Field of Search**..... 179/15 BC, 1 SA, 15.55 T, 179/15.55 R

[56] **References Cited**

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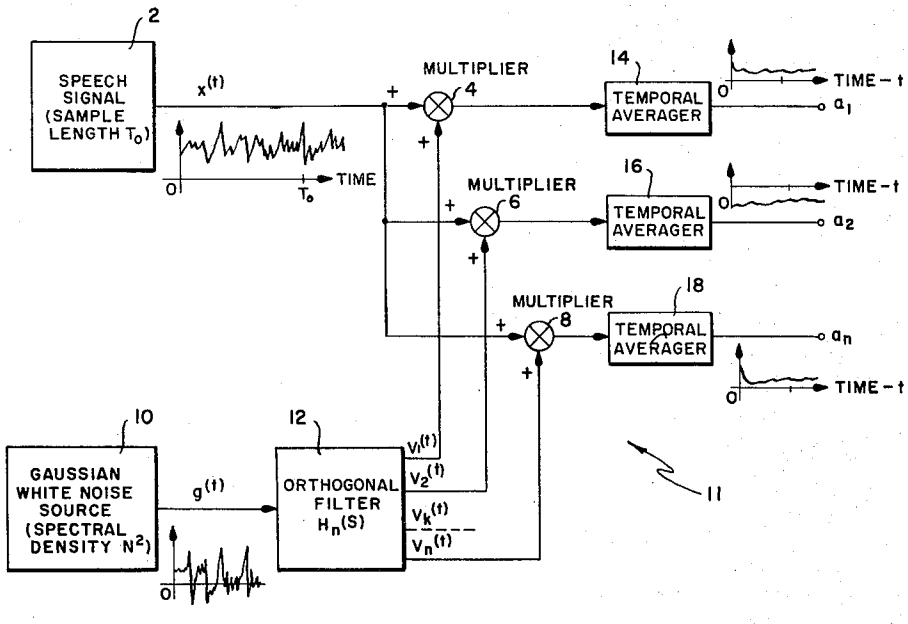
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Primary Examiner—Ralph D. Blakeslee
Assistant Examiner—Bradford Leaheey
Attorney, Agent, or Firm—R. S. Sciascia; Q. E. Hodges

[57] **ABSTRACT**

The generation of a set of a certain kind of coefficients related to the speech signal is obtained by simultaneously operating upon a sample of the speech signal with a set of orthogonal random functions derived from a white noise source and the white noise itself. These coefficients are sequentially AM or FM modulated for transmission by any suitable means. The transmitted signal is received and demodulated and sequentially processed through the system of this invention to recover the coefficients which are related to and can be used with a second source of white noise to reconstruct the original speech signal. The speech compression is accomplished by utilizing the coefficients as the information carrier source and later at the receiving ends to reconstruct the speech signal from these coefficients to obtain the vocal intelligence.

10 Claims, 4 Drawing Figures



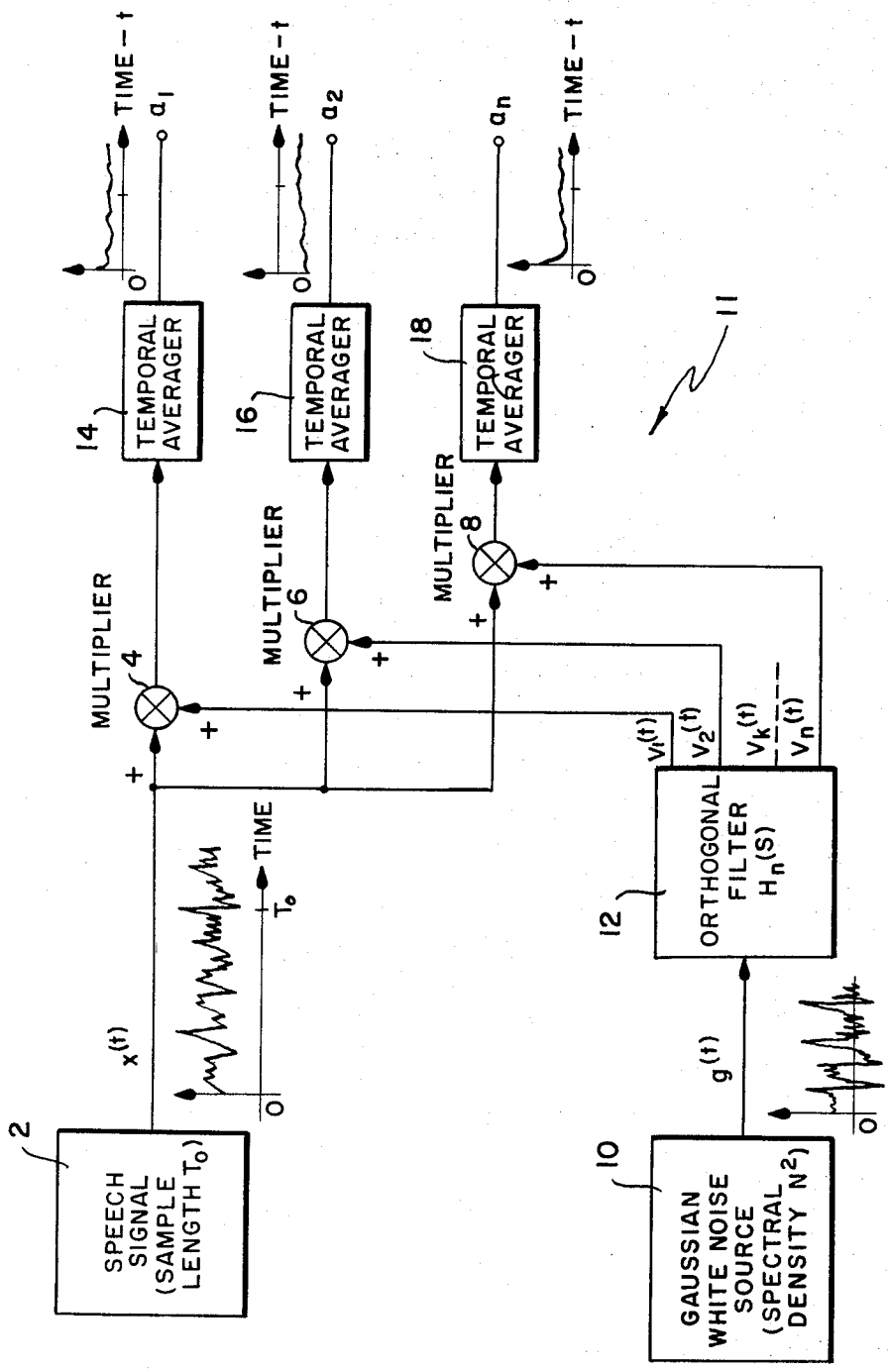


FIG. 1.

SYSTEM FOR OBTAINING THE SPEECH EXTRACTION COEFFICIENTS

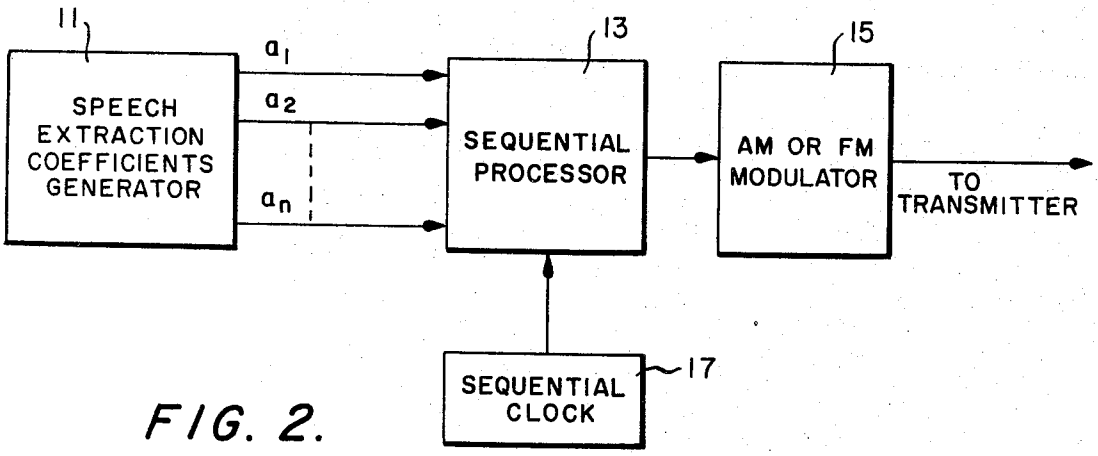


FIG. 2.

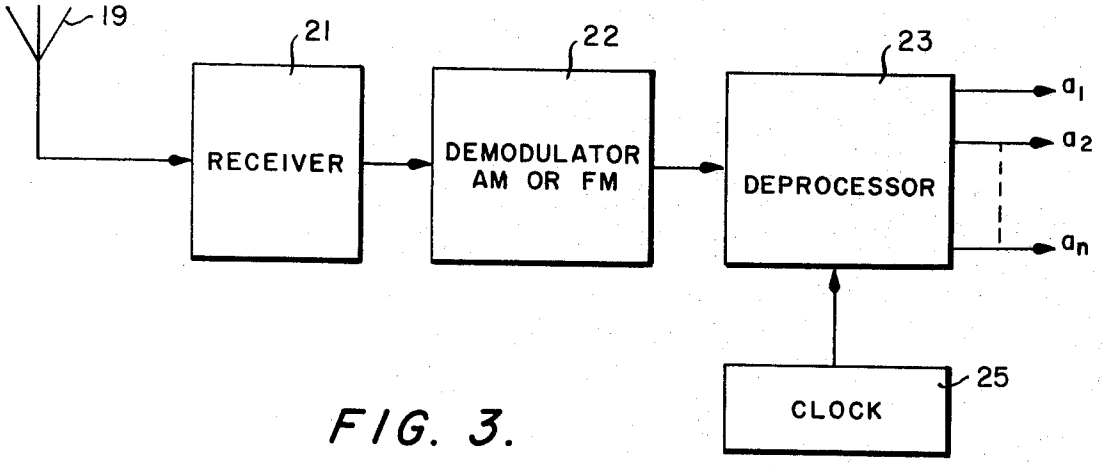


FIG. 3.

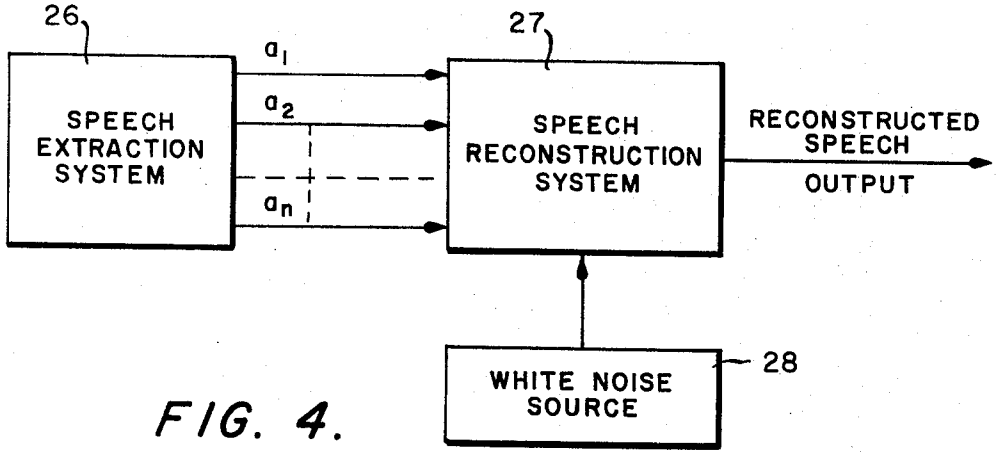


FIG. 4.

SPEECH COMPRESSION SYSTEM

This application is a continuation -in-part of application Ser. No. 191,003, filed Oct. 21, 1971 which was a continuation -in-part of application Ser. No. 155,988, filed June 23, 1971.

The invention described herein may be manufactured and used by or for the Government of the United States of America for governmental purposes without the payment of any royalties thereon or therefor.

BACKGROUND OF THE INVENTION

The increasing demands of the commumications industry for ways to transmit information and the crowding of the radio frequency spectrum, especially of speech conveyed information, makes it highly desirable to compress speech into as narrow a bandwidth as possible while transmitting the full range of information. This also decreases the power requirements to transmit a given piece of information.

SUMMARY OF THE INVENTION

This invention involves the application of the structure and techniques of the parent applications to the particular problem of speech compression. By utilizing the speech coefficients as information carrying elements of the transmitted signal it is possible to drastically narrow the bandwidths in comparison with conventional modes of transmitting information, the bandwidth required to transmit the required information, thereby accomplishing a significant speech compression.

It is an object of this invention to provide a system of speech transmission which reduces the bandwidth required to transmit speech signals.

Still another object of this invention is to transmit speech related signals which permit the reconstruction of the original speech.

It is also an object of this invention to provide a system including means to reduce speech to a set of speech coefficients which can be transmitted and utilized by a second means to reconstruct the original speech.

DESCRIPTION OF THE DRAWINGS

FIG. **1** is a diagram of the means for obtaining a set of speech extraction coefficients from speech.

FIG. **2** is a diagram of a system for processing the speech extraction coefficients for transmittal.

FIG. **3** is a diagram of a system for receiving and deprocessing a signal transmitted by radio from the system of FIG. **2.**

FIG. **4** is a diagram of a system for reconstructing speech from a set of speech extraction coefficients.

DESCRIPTION OF A PREFERRED EMBODIMENT

Referring now to FIG. **1,** where the system is generally designated as **11,** there is shown a source of speech signal having a sample length T_0. This signal which may be represented as a function of time, $x(t)$, is multiplied simultaneously in multipliers **4, 6, ...8,** etc. by a set of random orthogonal functions of time, $V_n(t)$ where $n = 1, 2, 3...$, generated from a white noise source **10** by the application of a set of orthogonal filters such as are disclosed in application Ser. No. 191,003. The number of multipliers and outputs from the orthogonal filters

being determined by the degree of fidelity required in the reproduction of the speech signal or the numbers of samples required to transmit the information. Each of the multiplier outputs is processed through its respective temporal averager **14, 16, 18** such as described in the referenced applications to generate as outputs the speech extraction coefficients $a_1, a_2 ..., a_n$. A speech extraction coefficient is defined herein as the temporal average of the product of a sample function from a set of orthogonal noise sample functions and a speech signal sample function. The operation of this system is more fully described in copending applications Ser. Nos. 191,003 and 155,988.

These speech extraction coefficients can be thought of as a set of generalized Fourier coefficients $a_1, a_2, ...a_n$ which are slowly varying time functions given by the following equations:

$$a_1=\frac{1}{T}\int_0^T V_1(t)X(t)dt$$

$$a_2=\frac{1}{T}\int_0^T V_2(t)X(t)dt$$

$$a_n=\frac{1}{T}\int_0^T V_n(t)X(t)dt$$

in which $X(t)$ is a sample of speech of length T_o and $V_n(t)$; $n = 1, 2, 3...$ are the set of outputs from the orthogonal filters in response to white noise, shown in block **12** of FIG. **1.** Since these coefficients are very slowly varying time functions they have very narrow bandwidth requirements. For example, the bandwidth for containing these functions may be of the order of 5HZ.

Referring now to FIG. **2,** the system of FIG. **1** is designated as block **11** the outputs of which are fed to a sequential processor which is a device driven by a clock **17** for sequentially gating the speech extraction coefficients to a modulator circuit **15** which modulates a carrier signal for transmission by any suitable means such as radio, telephone, light beam, etc. Such modulation may be either amplitude or frequency modulation.

The transmitted modulated signal is received by a system such as shown in FIG. **3,** for radio signals. The signal transmitted by the system of FIG. **2** is received by antenna **19** and fed to receiver **21,** both of which may be of conventional design. The received signal is appropriately demodulated in block **22** and sequentially processed by the Sequential Deprocessor **23** which is driven by clock **25** to reproduce the speech extraction coefficients $a_1, a_2...a_n$. These coefficients are utilized in a speech reconstruction system shown in FIG. **4** as block **27** to operate on the output of a source of white noise, **28** to reconstruct the original speech.

In FIG. **4** the block **26** may be the receiver system of FIG. **3** or any suitable receiving means for the transmitted speech extraction coefficients. For a more detailed description of the speech reconstruction system, reference is made to FIG. **5** of application Ser. No. 155,988 and the description in the specification of that application relating thereto.

The system of this invention utilizes the invention of application Ser. No. 191,003 now U.S. Pat. No. 3,800,093 to generate a set of speech coefficients which are used to modulate a suitable transmission carrier and the invention of application Ser. No. 155,988

now U.S. Pat. No. 3,746,791 to reconstruct the speech.

Obviously many modifications and variations of the present invention are possible in the light of the above teachings. It is therefore to be understood that within the scope of the appended claims the invention may be practiced otherwise than as specifically described.

What is claimed is:

1. A speech compression system including:

first means to receive a speech signal and to generate a set of speech extraction coefficients from said speech signal;

second means connected to said first means for transmitting said set of coefficients, said second means having an output;

third means for receiving the output of said second means;

fourth means connected to said third means for reconstructing a speech signal corresponding to the speech signal received by said first means;

whereby the bandwidth required to transmit each said speech extraction coefficient is narrower than the bandwidth required to transmit said speech signal.

2. The system of claim **1** in which the first means includes an orthogonal filter.

3. A system as claimed in claim **2** in which said coefficients are very slowly varying functions of time.

4. A system as claimed in claim **3** in which the time function of the coefficients are given by the equations

$$a_1=\frac{1}{T}\int_0^T V_1(t)X(t)dt$$

$$a_2=\frac{1}{T}\int_0^T V_2(t)X(t)dt$$

$$a_n=\frac{1}{T}\int_0^T V_n(t)X(t)dt$$

where $X(t)$ is a sample of speech signal of said length T_o, $V_n(t)$; $n = 1, 2, 3...$ are the set of outputs of an orthogonal filter in response to white noise and a_n; $n = 1, 2, 3...$are said set of coefficients.

5. A system as claimed in claim **2** in which said third means includes an orthogonal filter, said orthogonal filter of said third means having the same transfer characteristic as said orthogonal filter of said first means.

6. A system as claimed in claim **1** in which said first means includes a white noise source of spectral power density N^2 watts per cycle.

7. A system as claimed in claim **6** in which said third means includes a source of white noise of a spectral power density $1/N^2$ watts per cycle.

8. A method of communication and speech compression comprising the steps of:

producing a set of speech extraction coefficients from a speech signal such that each said coefficient has narrower transmission bandwidth requirements than said speech signal;

transmitting said speech extraction coefficients;

receiving said speech extraction coefficients;

reconstructing said speech signal from said speech extraction coefficients.

9. The method of claim **8** wherein the step of transmitting said speech extraction coefficients includes the step of sequentially processing said coefficients.

10. The method of claim **9** wherein the step of receiving said coefficients includes the step of sequentially deprocessing said coefficients.

* * * * *

UNITED STATES PATENT OFFICE

CERTIFICATE OF CORRECTION

Patent No. 3,833,767 Dated September 3, 1974

Inventor(s) Alfred A. Wolf

It is certified that error appears in the above-identified patent and that said Letters Patent are hereby corrected as shown below:

The term of this patent subsequent to

July 17, 1990, has been disclaimed.

Signed and sealed this 20th day of May 1975.

(SEAL)
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