Jan. 27, 1970 M. R. SCHROEDER 3,492,429

INTERPOLATION OF DATA WITH CONTINUOUS SPEECH SIGNALS

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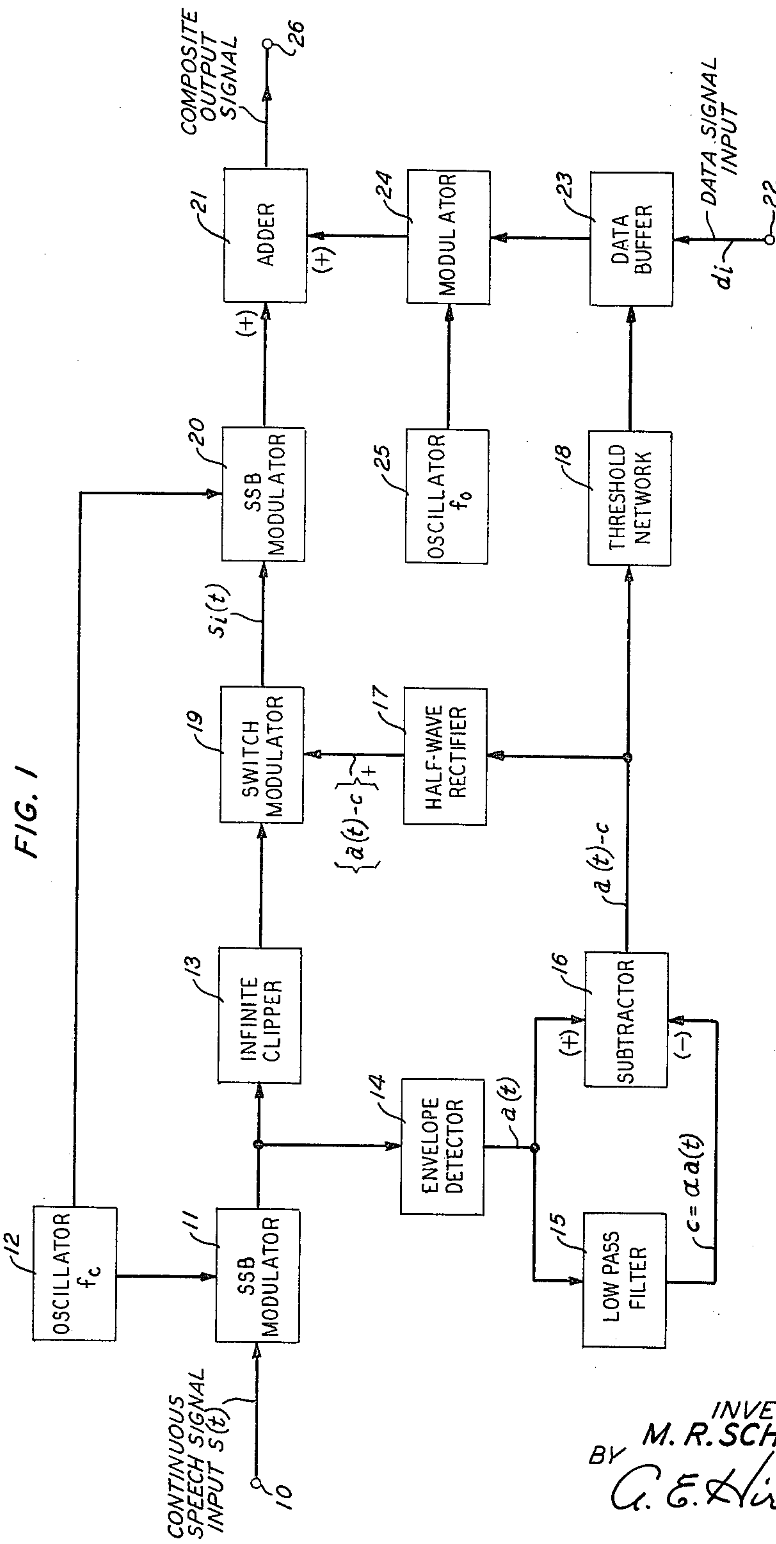
FIG. 1



INVENTOR
M. R. SCHROEDER
BY *A. E. Hirsch Jr.*
ATTORNEY

FIG. 2

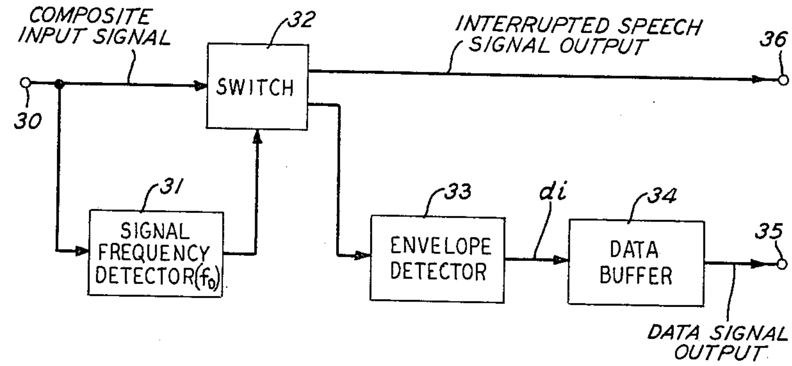


FIG. 3

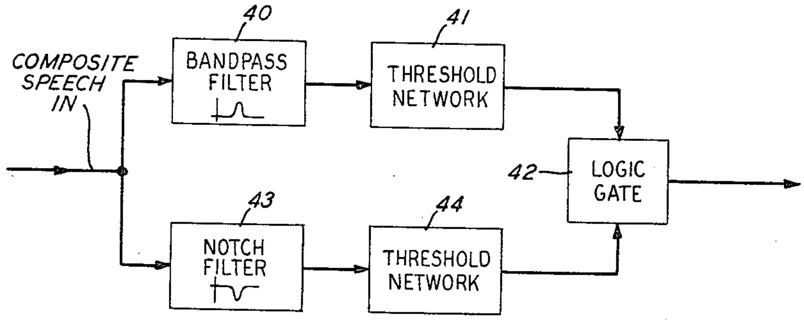


FIG. 4

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INTERPOLATION OF DATA WITH CONTINUOUS SPEECH SIGNALS

Manfred R. Schroeder, Gillette, N.J., assignor to Bell Telephone Laboratories, Incorporated, Murray Hill and Berkeley Heights, N.J., a corporation of New York

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U.S. Cl. 179—15 12 Claims

ABSTRACT OF THE DISCLOSURE

In communications systems in which several signals are interpolated for transmission according to prescribed time assignments, interruptions created in a signal, especially a speech signal, introduce distortions that generally reduce the quality of the signal. Degradation is due primarily to anharmonic components developed because of severe signal discontinuities. These undesirable effects may be overcome by interrupting speech at prescribed pitch synchronous intervals. Gaps in a speech signal thus are created at irregular intervals selected to minimize discontinuity and anharmonic effects.

BACKGROUND OF THE INVENTION

This invention relates to communication systems and, more particularly, to multiplex transmission systems in which both voice communication and data signals are transmitted over a single communication channel.

Field of the invention

In some communications systems, the need arises for transmitting speech signals and coded data over a single transmission channel. Multiplex systems permit this to be done. They generally have followed one of two patterns; either an allocation of signals to selected frequencies or to selected times. If a restricted bandwidth service is adequate, the available transmission band may thus be allocated on the frequency scale for two or more separate signal trains. Simultaneous data and speech transmission may thus take place continuously with each signal necessarily restricted to only a portion of the available channel bandwidth. Alternatively, if full-time service is not necessary, data signals may be interspersed during gaps in the speech, either in natural silent intervals or in gaps created by chopping. The first technique, though permitting simultaneous, continuous service, limits the quality and quantity of data and speech that can be transmitted. The second, though permitting full bandwidth transmission, is discontinuous and may severely degrade the quality of the speech signal.

Description of the prior art

Transmission systems in which data or signaling information is interspersed in a continuous speech signal are effective because a normal speech signal contains many gaps and pauses as well as listening intervals completely devoid of speech. Data may be transmitted during these gaps and pauses.

To increase the flow of data, it has been proposed that the speech signal be interrupted for the insertion of data at other than natural moments of speech silence. If the required gaps for data are created simply by interrupting the speech signal instead of waiting for intervals of silence, severe degradation of speech quality and some loss in intelligibility results. Of course, if this form of degradation can be tolerated, or is held to a low level, a greater allocation of data to the channel may be made.

The primary reason for the speech degradation which results from speech interruption or chopping is twofold:

(1) the interruptions introduce discontinuities in the speech signal, two for every interruption; and (2) the interruptions, unless they fortuitously occur at pitch synchronous intervals, create an inharmonic signal. Even if the timing intervals are judiciously selected, for example, by seeking times of low signal amplitude, a reproduction of the inharmonic signal is distorted. Moreover, rather complex signal analyzing equipment is required to seek out appropriate chopping times, with only a slight chance of improvement over purely random chopping. Obviously, the extent of the gaps carved out of the speech signal play an important role in determining the naturalness of the reconstituted speech signal.

Although the reconstitution of discontinously transmitted data presents no real problem since suitable buffering techniques are known, the reconstitution of speech presents a difficult problem. Mere play-back of speech signals with random gaps yields highly distorted speech. Play-back with interpolated speech to fill in the gaps is efficient but complex. Preferably, direct play-back of the interrupted speech signal with low distortion is desirable.

SUMMARY OF THE INVENTION

The outlined difficulties are overcome in accordance with the present invention by interrupting a continuous speech signal at prescribed times such that degradation of the signal is appreciably reduced. Reproduced speech is thus fully intelligible and of high quality.

It is therefore an object of this invention to improve the manner of interrupting a relatively continuous speech signal in order to permit data or other speech signals to be interspersed for multiplex transmission.

In accordance with the invention, a speech signal is interrupted at a pitch synchronous rate to produce gaps at irregular intervals in time. The gaps are available for the transmission of data or other speech; yet successive speech signal discontinuities "match" one another so that severe discontinuities are avoided. According to the invention, an analysis of the speech signal is employed to select those intervals for which the envelope of the speech signal is below a prescribed threshold. To do this, a signal representative of an envelope which differs from the envelope of the applied signal is impressed, for example, on a representation of the instantaneous phase of the applied signal, such that the modified instantaneous phase signal is diminished to a selected low level at desired intervals. An interrupted speech signal is reconstituted from the resultant. The intervals are also detected and employed, for example, for transferring signals from a buffer to the transmission channel. The buffer is employed to enable a continuous input flow of data to be stored temporarily, and read out intermittently at the times that gaps are available. Thus data signal segments are interpolated with the interrupted speech signal for transmission. At a receiver station the data signals are detected, for example, by noting the carrier frequency of the received signal, and separated from the composite transmitted signal.

DESCRIPTION OF THE DRAWINGS

The invention will be fully apprehended from the following detailed description of the illustrative embodiments thereof taken in connection with the appended drawings in which:

FIG. 1 is a block schematic diagram of a multiplex transmitter station constructed in accordance with the invention;

FIG. 2 is a block schematic diagram of a typical receiver station constructed in accordance with the invention;

FIG. 3 is a block schematic diagram of a data signal detector which may be used in the receiver apparatus of FIG. 2; and

FIG. 4 is a waveform diagram illustrating the development of a suitable switching function in accordance with the invention.

DETAILED DESCRIPTION

A pitch synchronous switching function for optimumly interrupting a speech signal for the insertion of data is developed in accordance with the invention by establishing a selected threshold for speech signals and pitch synchronously interrupting the signal when its amplitude falls below the established threshold. Thus, the average off-time ratio of the speech signal is a function of the established threshold. Regardless of the ratio, however, interruption takes place at prescribed instants such that discontinuities in the remaining speech signal are harmonically related, thus minimizing audible degradation.

The switching function is derived in the following manner. An applied speech signal,

$$s(t)=a(t)\cos[\varphi(t)] \quad (1)$$

is divided by its envelope,

$$a(t)=[s^2(t)+\hat{s}^2(t)]^{1/2}, \quad (2)$$

where $\hat{s}(t)$ is the Hilbert transform of $s(t)$. The resultant quotient signal $s(t)/a(t)$ is thereupon multiplied by a modified envelope signal derived from the original envelope $a(t)$. The modified envelope signal is developed from the original envelope by subtracting from it a constant proportional to the envelope and subjecting the difference to half-wave rectification. The constant c subtracted from the envelope $a(t)$ determines the extent of the gaps in the speech signal $s(t)$; it determines the threshold of switching and the off-time ratio. The modified envelope signal,

$$\begin{aligned}\tilde{a}(t)&=\{a(t)-c\}_+ \quad (3)\\ &=a(t)-c \text{ if } a>c,\\ &=0 \text{ if } a<c,\end{aligned}$$

where the function $\{\ \}_+$ equals its argument for positive arguments and is zero otherwise. The product of the quotient signal $s(t)/a(t)$ when multiplied by the modified envelope $\tilde{a}(t)$ yields the desired interruption signal,

$$s_i(t)=\frac{s(t)}{a(t)}\{a(t)-c\}_+ \quad (4)$$

As noted, the average off-time ratio depends on the magnitude of the constant c. For Gaussian signals, the ratio is given by

$$r_{\text{off}}=1-\exp[-c^2/a^2] \quad (5)$$

Typically, c is made equal to $0.5\bar{a}$. For a Gaussian signal, this choice corresponds to

$$c=1/4[\pi\overline{a^2}]^{1/2} \quad (6)$$

Thus, the average off-time becomes

$$r_{\text{off}}=1-\exp[-\pi/16]=0.18=18\% \quad (7)$$

Actual observed off-time ratios for $c=0.5\bar{a}$ are approximately 26% for male speech and 16% for female speech. In this observation, $\bar{a}$ was obtained by averaging $a(t)$ over 20 msec. with a rectangular time window.

FIG. 1 illustrates apparatus for carrying out the operations defined mathematically above. A continuous speech signal $s(t)$ for example, of the form illustrated in FIG. 4(a), is applied to the transmitter apparatus of FIG. 1, for example, at input terminal **10**. For implementation simplicity, the signal is modulated on a carrier frequency, f_c, by passing it through single sideband modulator **11** which is supplied with carrier signal oscillations at frequency f_c from oscillator **12**. The resulting modulated signal is thereupon delivered both to infinite clipper **13** and to envelope detector **14**. The single sideband modulation step together with infinite clipping yields, in a fashion well known to those skilled in the art, the quotient $s(t)/a(t)$. Envelope detector **14** yields at its output the envelope of $s(t)$, viz., $a(t)$. The envelope signal is passed through low-pass filter **15** to produce a constant signal $c=\alpha a(t)$. The constant signal c is thereupon subtracted from envelope signal $a(t)$ in subtractor **16** to produce a difference signal $a(t)-c$, which is delivered both to half-wave rectifier network **17** and threshold network **18**. Half-wave rectification of the quotient signal produces the desired switching function $\{\ \}_+$. A typical switching signal is illustrated in FIG. 4(b). The signal is used to control the action of switch modulator **19**, supplied at its signal input with the quotient signal $s(t)/a(t)$. The resulting product is equal to the desired switching function $s_i(t)$ as defined in Equation 4 and illustrated in FIG. 4(c). It is reduced to baseband by application to single sideband modulator **20** which is supplied with carrier signals at frequency f_c from oscillator **12**. The resulting discontinuous speech signal,

$$\{a(t)-c\}_+\cos[\varphi(t)] \quad (8)$$

is applied to one input of adder **21**.

Data signals d_i of any desired sort are applied to the terminal apparatus of FIGURE 1, for example at terminal **22**. Because of the discontinuous nature of their transmission, the applied data signals are passed through data buffer **23**. In typical fashion, data buffer **23** absorbs applied continuous signals, stores them, and makes them available upon call at irregular times. The transfer of data signals from the buffer to modulator **24** is under control of signals supplied from threshold network **18**. Thus, data is transferred out of the buffer only when the different signal $a(t)-c$ is below a selected ratio factor. The threshold of network **18** is thus set in order that control signals reach buffer **23** when the envelope of the continuous input signal is arbitrarily small and at pitch synchronous chopping points. Data signals from buffer **23** are thus delivered to modulator **24**. Modulator **24** is supplied with carrier signals at frequency f_0 from oscillator **25**, to produce a modulated output signal, $d_i \cos(2\pi f_0 t)$. The modulated signal is supplied to the second input terminal of adder **21**, and there interleaved in the interrupted speech signal supplied from modulator **20**. Since the interruptions in the speech occur synchronously with the delivery of data, a substantially continuous output signal is produced. The composite output signal may then be transmitted from terminal **26** in any desired fashion to a receiver station.

A suitable receiver terminal for composite data and speech signals is shown in FIGURE 2. Composite signals received at terminal **30** are examined in single frequency detector **31** for carrier signals at frequency f_0. The presence of this signal indicates an interval of data transmission. Accordingly, switch **32** is actuated to deliver the composite input signal from terminal **30** to envelope detector **33**. The detector, which may be of any conventional type, removes the carrier f_0 and delivers a baseband data signal to data buffer **34**. In conventional fashion, data is accumulated in the buffer and, as desired, delivered as a continuous data output signal, for example, to terminal **35**, and thereupon to a data utilization device, not shown.

In the absence of a detected signal at frequency f_0, switch **32** supplies input signals from terminal **30** directly to speech output terminal **36** for delivery to a speech utilization device, not shown. Although output signals available at terminal **36** are interrupted. the interruptions, as noted above, are insignificant insofar as speech intelligibility is concerned since the discontinuities bounding the gaps are small and harmonically related. Hence, the speech signal may be used directly with minimum perceivable degredation.

A suitable single frequency detector for use in the receiver apparatus of FIGURE 2 is shown in FIGURE 3. The composite input signal is passed through baseband filter **40**, centered about frequency f_0, in order that a signal component at frequency f_0, if present, is delivered to threshold network **41**. Network **41** responds only to signals above a selected threshold θ to produce a control signal. Thus, if a component at frequency f_0 of substantial magnitude is present, indicating that a carrier signal from oscillator **25** at the transmitter is present in the composite signal, as control signal is produced by network **41** and delivered to logic gate **42**. The composite signal is also supplied to notch filter **43** centered about frequency f_0 in order to block components at that frequency. Thus, if a speech or noise signal component at frequency f_0 happens to be present in the composite signal (so as to reach threshold network **41**), it is blocked by filter network **43**. However, other speech or noise components which accompany the component at f_0 are passed by filter **43** and, if of sufficient magnitude, exceed the selected threshold of network **44**. A control signal is thereupon delivered by network **44** to logic gate **42**. Thus, network **44** provides an additional control signal for logic gate **42** to improve detection reliability. If the network of FIGURE 3 detects a carrier signal at frequency f_0, logic gate **42** supplies the appropriate control signal to switch **32** (FIGURE 2) for routing incoming signals to the data channel. If the network of FIGURE 3 fails to detect a carrier component, the signal from gate **42** actuates switch **32** to connect the input terminal to the speech channel.

The above-described arrangements are, of course, merely illustrative of the application of the principles of the invention. Numerous other arrangements may be devised by those skilled in the art without departing from the spirit and scope of the invention.

What is claimed is:

1. Speech signal processing apparatus, which comprises,
   means supplied with a speech signal for developing a signal representative of the instantaneous phase of said speech signal,
   means for determining intervals for which said instantaneous phase signal is below a prescribed threshold, and
   means for pitch synchronously interrupting said speech signal during said selected intervals.

2. In combination,
   means for developing a signal representative of the instantaneous phase of an applied speech signal,
   means for impressing on said instantaneous phase signal a modified envelope which differs from the envelope of said applied speech signal by an essentially constant amount, such that the magnitude of said modified phase signal is diminished substantially to zero at selected intervals,
   means responsive to intervals of substantially zero magnitude in said modified signal for interrupting said applied speech signal, and
   means for utilizing said interrupted signal.

3. Apparatus for developing a pitch synchronous switching function for interrupting a speech signal to create intervals for the interpolation of other signals, which comprises,
   means supplied with a continuous speech signal for developing a signal representative of the envelope of said speech signal,
   means for developing a modified envelope signal, said means including means for subtracting from said envelope signal a constant signal derived from said envelope signal to produce a difference signal, and
   means for rectifying said difference signal.

4. Apparatus for interrupting a speech signal to create intervals for the interpolation of other signals, which comprises,
   means supplied with a continuous speech signal for developing a signal representative of the instantaneous phase of said speech signal,

   means for developing a signal representative of the quotient of said speech signal divided by said phase representative signal,
   means supplied with said continuous speech signal for developing a signal representative of the envelope of said speech signal,
   means for developing from said envelope signal a modified envelope signal, said means including means for subtracting from said envelope signal a constant signal derived from the average of said envelope to produce a difference signal,
   means for rectifying said difference signal, and
   means for developing a signal proportional to the product of said quotient signal and said modified envelope signal.

5. Apparatus for the multiplex transmission of a speech signal and other signals, which comprises:
   means supplied with a continuous speech signal for developing a signal representative of the phase of said speech signal;
   means for developing a signal representative of the quotient of said speech signal divided by said phase signal;
   means for developing a signal representative of a modification of the envelope of said continuous speech signal, said means including means for subtracting from an envelope representative signal a constant derived from said envelope to produce a difference signal, and means for rectifying said difference signal;
   means for developing a signal proportional to the product of said quotient signal and said modified envelope signal;
   means for determining intervals during which the magnitude of said difference signal is below a selected threshold;
   means responsive to said determined below-threshold difference signals for selecting intervals of said other signals for transmission; and
   means for combining said selected intervals of said other signals with said product signal for transmission.

6. A multiplex transmission system in which idle gaps created in a speech signal are utilized for the transmission of other signals, which comprises,
   means for developing a signal representative of instantaneous phase of an applied speech signal,
   means for modulating said phase signal by a selected signal derived from the envelope of said applied speech signal of a magnitude sufficient to diminish the instantaneous magnitude of said phase signal substantially to zero at selected intervals,
   means for developing an interrupted signal from said diminished phase signal,
   means responsive to intervals of substantially zero amplitude in said envelope derived signal for selecting segments of said other signal for transmission, and
   means for interpolating interrupted speech signal and said selected other signal segments for transmission.

7. A multiplex transmission system in which idle gaps created in a speech signal are utilized for the transmission of data signals, which comprises: at a transmitter station;
   means for developing a signal representative of the instantaneous phase of an applied speech signal,
   means for impressing on said phase signal an envelope signal sufficient to diminish the instantaneous magnitude of said phase signal to zero at selected intervals, thereby develop an interrupted speech signal,
   a source of a data signal,
   means responsive to intervals of zero magnitude in said phase signal for selecting data signal segments for transmission,

means for interpolating said interrupted speech signal and said selected data signal segments for transmission,

means for transmitting said interpolated signal to a receiver station; and, at said receiver station;

means for identifying intervals in said interpolated signal which represent said interrupted speech signal and intervals which represent segments of said data signal, and

means responsive to said identifying means for separating said speech and data signals.

8. Apparatus for interrupting a speech signal for transmission, which comprises:

means supplied with an input speech signal $s(t)$ and with a carrier signal of frequency f_c for shifting the frequency of said speech signal $s(t)$;

means for infinitely clipping said frequency shifted speech signal to produce a quotient signal proportional to $s(t)/a(t)$, where $a(t)$ represents the envelope of said signal $s(t)$;

detector means for developing a signal $a(t)$ proportional to the envelope of said signal $s(t)$;

filter means supplied with said envelope signal $a(t)$ for developing a control signal $c=\alpha a(t)$;

means for subtracting said control signal c from said envelope signal $a(t)$ to produce a difference signal $a(t)-c$;

means for rectifying said difference signal to produce a modified envelope signal $\{a(t)-c\}_+$, which is equal to its argument for positive arguments and is otherwise zero;

means for modulating said quotient signal $s(t)/a(t)$ by said modified envelope signal $\{a(t)-c\}_+$ to produce an interrupted speech signal

$$s_i=s(t)/a(t)\cdot\{a(t)-c\}_+$$

means supplied with said interrupted signal s_i and with carrier signals at frequency f_c to produce an interrupted speech signal shifted in frequency to baseband, and

means for utilizing said interrupted speech signal for transmission.

9. In combination with apparatus as defined in claim **8**:

means for determining intervals for which the amplitude of said difference signal $a(t)-c$ is below a selected threshold level;

buffer means supplied with a data signal d_i for storing said data signal;

means responsive to said determined intervals for transferring said data signal from said buffer;

means for modulating said transferred data signal with a carrier signal at frequency f_0;

means for combining said modulated data signal with said interrupted speech signal; and

means for transmitting said combined signals to a receiver station.

10. In combination with apparatus as defined in claim **9**:

a receiver station which includes;

means for identifying intervals in received combined signal which represent said interrupted speech signal and intervals which represent said modulated data signal; and

means responsive to said identifying means for separating said speech and said data signals for independent utilization.

11. A multiplex transmission system as defined in claim **10**, wherein:

said means for identifying intervals in said combined signal which represent said interrupted speech signal and intervals which represent said modulated data signal, comprises:

bandpass filter means proportioned to pass signals modulated with carrier signals at frequency f_0; and

means for selecting signals passed by said bandpass filter means which exceed a prescribed threshold.

12. A multiplex transmission system a defined in claim **10**, wherein:

said means for identifying intervals in said combined signal which represent said interrupted speech signal and intervals which represent said modulated data signal, comprises:

bandpass filter means proportioned to pass signals modulated with carrier signals at frequency f_0;

first means for selecting signals passed by said bandpass filter means which exceed a first prescribed threshold;

band rejection filter means proportioned to reject signals modulated with carrier signals at frequency f_0;

second means for selecting signals passed by said band rejection filter means below a second prescribed threshold; and

logic gate means supplied with signals from said first and said second selection means for developing a control signal for signals above said first threshold and below said second threshold.

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RALPH D. BLAKESLEE, Primary Examiner

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