COMPRESSED FREQUENCY COMMUNICATION SYSTEM

Fig. 2

Fig. 3

Fig. 4

Fig. 5

Fig. 6

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COMPRESSED FREQUENCY COMMUNICATION SYSTEM


Application May 22, 1952, Serial No. 289,344

14 Claims. (Cl. 179—15)

This invention relates to a system for reducing the bandwidth requirements of a communication system and more particularly a system for facilitating the transmission of wide frequency band signals, such as speech, within a relatively narrow frequency band.

The information transmitted by speech does not, at any one time, necessarily require all the frequency space allotted to it by the human voice. The lack of utilization of the entire available spectrum all of the time is obvious when it is observed that any one person's speech is mostly a time sequence of voiced (periodic and unvoiced (aperiodic) sounds, each having different and restricted frequency spectra. Voiced sounds, including vowels and consonants like L, M, and N, involved the use of the vocal cords, while unvoiced sounds, such as T, K, and P, are produced in the mouth. In general, the significant energy of the voiced sounds occupies the lower portion of the frequency spectrum and contains a definite fundamental frequency while the significant energy of the unvoiced sounds occupies almost exclusively, the high portion of the audible frequency spectrum and has no fundamental frequency present.

The presence of two almost distinct bands of audible sounds has led to attempts to communicate over a frequency bandwidth corresponding essentially to the width of one of these bands, instead of the combined width of both. In the past the voiced and unvoiced sounds have both been passed through time invariant (static) filters causing the intelligence of the output speech to vary for different locations of the filter's center frequency. When the center frequency of the filter was located in the lower portion of the audio spectrum, the voiced sounds were best understood but the majority of the significant energy of the unvoiced sounds was lost, while when the center frequency of the invariant filter was placed in the upper portion of the spectrum, the unvoiced sounds were best understood and the voiced sounds were mutilated.

It is known that the maximum syllable and/or word articulation for a 1000 cycle transmission bandwidth is obtained with an invariant filter when the filter pass band contains the portion of the original speech spectrum lying between 200 and 1200 C. P. S., where most of the voiced sounds lie. This method effects frequency compression merely by placing the available transmission frequency bandwidth at the place in the original speech spectrum which results in maximum articulation for time stationary filters. However, this method of frequency compression loses the significant energy present in the majority of unvoiced sounds which lie in the spectrum outside the frequency pass band of such an invariant filter. The optimum 1000 cycle pass band for unvoiced sounds is located above 1.8 kc, but such a pass band would mutilate the voiced sounds. Accordingly, one of the principal objects of this invention is to provide an improved compressed frequency communication system utilizing separate time invariant filters for the voiced and unvoiced sounds of the original speech input.

Another object of this invention is to achieve frequency compression of speech by placing the available transmission bandwidth at the place in the speech spectrum which results in maximum articulation for the particular sound input for time stationary filters.

A further object of this invention is to provide a communication system capable of transmitting simultaneously a plurality of voiced currents without mutual interference and within a relatively narrow frequency bandwidth.

A feature of this invention is the use of a voiced-unvoiced sound detector to control the place of the available transmission bandwidth at the place in the speech spectrum which results in maximum articulation for the particular instantaneous speech input, then to relegate the separated passed frequency bands to the available portion of the transmission bandwidth and simultaneously transmit a frequency shift synchronizing signal to enable the receiver to demodulate the transmitted frequency bands of the original speech input.

Another feature of this invention is the multiplexing of a plurality of speech transmission channels by relegateing the filtered speech sounds of each channel to a predetermined portion of the transmission bandwidth and simultaneously transmitting a synchronizing signal for each channel to enable the receiver to demodulate the received speech currents of each channel so that only a relatively narrow frequency transmission bandwidth is required to conduct a plurality of independent speech messages.

The above-mentioned and other features and objects of this invention will become more apparent by reference to the following description taken in conjunction with the accompanying drawings, in which:

Fig. 1A is a schematic diagram in block form of a transmitter for use in a three-channel compressed frequency communication system;

Fig. 1B is a schematic diagram in block form of a receiver for use with the transmitter of Fig. 1A;

Fig. 2 is a schematic diagram in block form of one embodiment of a voiced-unvoiced detector for use with the transmitter of Fig. 1A;

Fig. 3 is a graphic illustration helpful in the explanation of the detector of Fig. 2;

Fig. 4 is a schematic diagram in block form of a second embodiment of a voiced-unvoiced detector for use with the transmitter of Fig. 1A;

Fig. 5 is a schematic diagram partly in block form of a third embodiment of a voiced-unvoiced detector; and

Fig. 6 is a schematic diagram in block form of an alternate embodiment of the detector of Fig. 5.

Referring to Fig. 1A, a multichannel compressed frequency communication system in accordance with the principles of this invention is shown comprising three speech channels, designated channel #1, channel #2, and channel #3. The original speech input to channel #1 is coupled through a speech amplifier 1 to a voiced-unvoiced detector 2 which determines if the instantaneous speech input comprises a voiced or unvoiced sound so that the available transmission bandwidth might be allocated to the most advantageous portion of the audio spectrum for the particular input sound. The output of the synchronous relay 3, responsive to the determination of the voiced-unvoiced detector 2, is coupled through delay line 4 to control speech relay 5. When the instantaneous input channel #1 is voiced, the output of the synchronous relay 3 causes the speech relay armature 6 to be moved to the upper or voiced position as shown. The output of speech amplifier 1 is coupled to a delay circuit 7 which delays the input signal an amount of time sufficient to allow the output of the voiced-unvoiced detector 2 and associate circuits to control the
speech relay 5. When the speech relay armature 6 is in the upper position, the output of the delay circuit 7 is fed directly to band pass filter 8 via line 9. Band pass filter 8 passes signals having a frequency between 2 and 12 kc, thus allowing the significant energy of the voice signals to be passed and coupled to mixing amplifier 10 over line 11. When the output of the synchronous relay 3 causes the speech relay armature 6 to be moved to the upper or voiced position, it also causes a frequency shift oscillator 12 to emit a 1350 C. P. S. synchronous signal which is passed through a synchronous band pass filter 14 to eliminate any spurious frequencies from the output of the oscillator 12. The filtered synchronous signal output of band pass filter 13 is also coupled to mixing amplifier 16 via line 11.

When the instantaneous speech input to channel #1 comprises unvoiced sounds, the output of the voice-unvoiced detector 2 causes the synchronous relay 3 output, delayed in circuit 4, to move the speech relay armature 6 to the lower or unvoiced position. When the armature 6 is in the lower position, the output of the speech amplifier 1 is coupled through delay circuit 7 to a band pass filter 14 passing signals between 1.8 and 2.8 kc, which is advantageous for passing the significant energy contained in unvoiced sounds. The frequency filtered energy is coupled to modulator 15. A separate unit 16 at the transmitter generates the frequency necessary for modulating the sampled unvoiced sounds for the portion of the transmitted frequency spectrum allocated to channel #1. The separate relay 16 comprises a 200 C. P. S. oscillator 17 whose output is distorted in frequency multiplier 18 to yield higher harmonics. The 3 kc. harmonic from frequency multiplier 18 is coupled to modulator 15 where it modulates the filtered 1.8 to 2.8 kc. unvoiced sound to yield a 1.2 to 2 kc. modulated unvoiced sound which is coupled through band pass filter 8 to mixing amplifier 10.

When the output of the voice-unvoiced detector 2 causes the output of synchronous relay 3 to move the armature 6 to the lower unvoiced position, the output of relay 3 causes the frequency shift synchronous oscillator to emit a 1290 C. P. S. synchronous signal which is filtered in synchronous band pass filter 13 which passes frequencies between 1275 and 1375 kc. and is coupled to modulator 10. Thus the output of the mixing amplifier 10, when the instantaneous sound input to channel #1 is voiced, comprises the 2 to 1.2 kc. frequency band of the voiced sound and a 1360 C. P. S. synchronous signal, but if the instantaneous input to channel #1 is unvoiced, the output of mixing amplifier 10 comprises the 1.8 to 2.8 kc. frequency band of unvoiced sounds modulated to the 1.2 to .2 kc. allocated frequency band of channel #1 and a 1290 C. P. S. synchronous signal.
The speech input to channel #2 is coupled through speech amplifier 19 to a voice-unvoiced detector 20. In a manner similar to the operation in channel #1, the output of the voice-unvoiced detector 20 is fed to a synchronous relay 21 and through a delay line 22 to operate the speech relay 23. The armature 24 of the speech relay 23 is moved to the upper position when the instantaneous speech input to channel #2 is voiced and to the lower position when the instantaneous speech input is unvoiced. When the instantaneous speech input is unvoiced and the armature 24 is in the lower position, the output of the speech amplifier 19 is coupled through a delay circuit 25, which imposes a delay equal in time to the time necessary for the speech relay 23 to a band pass filter 26, which passes only the significant energy lying in the 1.8 to 2.8 kc. frequency band which is the optimum frequency pass band for unvoiced sounds. This filtered energy is coupled via line 11 to mixing amplifier 10.

When the instantaneous speech input is composed of unvoiced sounds, the output of synchronous relay 21 causes the synchronous oscillator 27 to emit a 1465 C. P. S. synchronous signal which is coupled through band pass filter 28, passing frequencies between 1450 and 1550 C. P. S., to mixing amplifier 10. When the instantaneous speech input to channel #2 comprises voiced sounds and the speech relay armature 24 is moved to the upper position, the output of the speech relay 23 is fed to a delay circuit 29, which imposea a delay sufficient to the output of the voice-unvoiced detector 20 and associate circuits 21 and 22, the output of the speech amplifier 19 is coupled through delay circuit 25 to a low pass filter 29 which passes the optimum frequencies, below 1.2 kc., for voiced sounds. This frequency filtered energy from filter 29 is coupled through band pass filter 26 and mixing amplifier 10.

When the output of synchronous relay 21 causes the armature 24 to move to the upper position, it simultaneously causes the synchronous frequency shift oscillator 27 to emit a 1350 C. P. S. synchronous signal which is filtered in band pass filter 28 and coupled to mixing amplifier 10. In a manner similar to the operation of channels #1 and #2, the instantaneous speech input to channel #3 is coupled to a speech amplifier 31 whose output is fed to band pass filter 32 and voice-unvoiced detector 23. The output of synchronous relay 34, responsive to the determination of the speech characteristic by voice-unvoiced detector 33, is coupled through a delay line 35 to operate the speech relay 36. The speech input delayed in circuit 32 is coupled through a low pass filter 37 having an upper pass frequency of 3.2 kc. and coupled to modulator 38.

When the instantaneous speech input to channel #3 is voiced, as determined by voice-unvoiced detector 33, the armature 39 of speech relay 36 is moved to the upper or voiced position which couples a 4 kc. signal from frequency multiplier 18 to modulator 38. The 4 kc. signal output of frequency multiplier 18 is obtained by distortion of the 200 C. P. S. signal from oscillator 17, as was the 3 kc. signal utilized in channels #1 and #2. The 4 kc. signal is utilized in circuit 38 to modulate the speech output filter of filter 37 and the output is coupled through a 2.8 to 3.8 kc. band pass filter 40. Since filter 40 passes only the 2.8 to 3.8 kc. frequency to the modulator output, only the portion of the original speech input lying between .2 and 1.2 kc. modulated by the 4 kc. signal will be contained in the output of filter 40 and coupled via line 11 to mixing amplifier 10. The output of synchronous relay 34 causing the speech relay armature 39 to move to the upper position also causes synchronous oscillator 41 to emit a 1710 C. P. S. synchronous signal which is filtered in band pass filter 42 designed to pass frequencies between 1625 and 1725 C. P. S. The filtered synchronous signal is coupled to mixing amplifier 10 over line 11 with the modulated voice frequency components from filter 40.

When the output of the voice-unvoiced detector 33 causes armature 39 to move to the lower or unvoiced position, a 6 kc. modulating signal is coupled to modulator 38. The 6 kc. signal is obtained by doubling the 3 kc. output of frequency multiplier 18 in frequency doubler circuit 43 and coupling its output through armature 39 to modulator 38. The modulated speech output of modulator 38 is coupled to band pass filter 40 passing frequencies between 2.8 and 3.8 kc. The 2.8 to 3.8 kc. output of modulator 38 is equal to the frequencies of the speech input lying between 2.2 and 3.2 kc., which is an advantageous band for the significant energy of unvoiced sounds, modulated by the 6 kc. signal from frequency doubler 43.

When the output of the synchronous relay 34 causes the armature 39 to be moved to the lower position, it also causes the synchronous frequency shift oscillator 41 to
emit a 1640 C. P. S. synchronous signal which is filtered by band pass filter 42 and fed to mixing amplifier 10.

Thus the output of mixing amplifier 10 comprises either the .2 to 1.2 kc. frequency band of voiced sound input to channel #1 and a 1360 C. P. S. synchronous signal or the channel #1 modulated to the .2 to 1.2 kc. transmission frequency band allocation of channel #1 and a 1290 C. P. S. synchronous signal; and either the .2 to 2.8 kc. frequency band of unvoiced sound input to channel #2 and a 1465 C. P. S. synchronous signal or the .2 to 1.2 kc. frequency band of voiced sound input to channel #2 and a 1535 C. P. S. synchronous signal; and either the .2 to 1.2 kc. frequency band of voiced sound input to channel #3 modulated to the 2.8 to 3.8 kc. transmission frequency band allocation of channel #3 and a 1710 C. P. S. synchronous signal or the 2.2 to 3.2 kc. frequency band of unvoiced sound input to channel #3 modulated to the 2.8 to 3.8 kc. transmission frequency band allocation of channel #3 and a 1640 C. P. S. synchronous signal. The 200 C. P. S. basic modulating signal from oscillator 17 is also coupled to mixing amplifier 10. The output from amplifier 10 is coupled to the transmitter 44 which transmits all three speech channel currents and synchronizing signals within the frequency band of .2 to 3.8 kc.

The transmitted signals are detected in the usual receiver circuitry 45. Band pass filter 46 passes all received signals lying in the frequency band between .2 to 1.2 kc. which contains the transmitted speech currents of channel #1. The synchronous band pass filter 47 passes the synchronous signals of channel #1, either 1360 or 1290 C. P. S., to discriminator 48 whose output controls the movement of the armature 49 of speech relay 50. When the synchronous signal has a frequency of 1360 C. P. S., it indicates a voiced sound is being received in channel #1 and the armature 49 is moved to the upper voice position and causes the clipper discriminator 51 to operate. When the synchronous signal has a frequency of 1290 C. P. S., it indicates an unvoiced sound is being received in channel #1 and the armature 49 is moved to the lower unvoiced position coupling the output of filter 46 to modulator 52.

The 200 C. P. S. basic modulating signal utilized in the transmitter to obtain the synchronized signals for each channel is detected in receiver 45 and passed by filter 53 to a frequency multiplier circuit 54 which distorts the filtered basic modulating signal to provide the 3 kc, 4 kc., and 6 kc. harmonics in its output. This procedure provides modulating frequencies in the receiver from the basic modulating signal and allows the same frequency to be maintained at the modulator and demodulator for each channel in spite of slight drifts in the 200 C. P. S. oscillator 17 of the transmitter.

The 3 kc. frequency output of multiplier 54 is fed to modulator 52 where it is combined with the .2 to 1.2 kc. received signal to demodulate the signal to its original position in the frequency spectrum, e. g. 1.8 to 2.8 kc. The demodulator unvoiced sound energy is coupled to band pass filter 55 to remove any spurious responses due to the demodulating step and the filtered output is coupled to variable resistor 51. The output of resistor 51 is coupled through tap 56 to amplifier 57 whose output is equal to the reconstructed speech input to channel #1 of the receiver. The tap 56 of variable resistor 51 is adjusted so that the output of receiver channel #1 contains a pleasing balance of voiced to unvoiced sounds.

In a similar manner the 1.8 to 2.8 kc. signals representing the transmitted speech input of channel #2 are filtered by speech band pass filter 58 of receiver channel #2. The synchronous signals of channel #2 are passed by the synchronous band pass filter 59 and when the received synchronous signal is 1535 C. P. S., the output of discriminator 60 causes the speech relay armature 61 to be moved to the voiced sound position where the output of the speech band pass filter 58 is coupled to the modulator 62. The output of the speech band pass filter 58 is beat with the 3 kc. modulating signal obtained from frequency multiplier 54 and the .2 to 1.2 kc. output of modulator 62 is passed through a low pass filter 63 to the variable resistor 64 of channel #2. When the synchronous band pass filter 59 passes a synchronous signal of 1465 C. P. S., the output of the discriminator 60 causes the speech relay armature 61 to be moved to the unvoiced position where the 1.8 to 2.8 kc. signal is coupled directly to the variable resistor 64 of channel #2. The balanced voiced-unvoiced output of the variable resistor 64 is coupled through tap 65 to amplifier 66 to the output circuitry and comprises the reconstructed original speech input to channel #2.

The received signals lying in the 2.8 to 3.8 kc. frequency band of channel #3 are filtered by the speech band pass filter 67 and passed to the modulator 68 of channel #3. When the synchronous band pass filter 69 passes a 1710 C. P. S. synchronous signal, the output of the discriminator 70 causes the speech relay armature 71 to be moved to the upper position which couples the 4 kc. output of the frequency multiplier 54 to modulator 68 where it is beat against the 2.8 to 3.8 kc. output of the speech band pass filter 72. The output of modulator 68 for the voiced position of the speech relay armature 71 comprises a .2 to 1.2 kc. demodulated output which is passed to the low pass filter 73 and coupled to amplifier 73 and thence to the speech output of channel #3. When the synchronous signal for channel #3 comprises a 1640 cycle signal, the output of the discriminator 70 causes the speech relay armature 71 to be moved to the lower unvoiced position where the 6 kc. output of the frequency multiplier 54 is coupled to the modulator 68 to be beat against the 2.8 to 3.8 kc. output of the speech band pass filter 72. The demodulated 2.2 to 3.2 kc. signal from the modulator 68 is passed through the low pass filter 72 to the amplifier 73. The output of the amplifier 73 is the reconstructed speech input to channel #3 of the transmitter. Thus the output of the receiver circuitry comprises the reconstructed separated speech input to each of the three audio channels of the transmitter.

One type of detector, for use in the communication system of this invention, for determining whether the instantaneous speech input to any channel is voiced or unvoiced, is shown in Fig. 2. The amplified instantaneous speech input of the channel is coupled through a volume compressor 74 to two filters 75 and 76, one band pass at 350 to 900 C. P. S. and the other high pass above 3500 C. P. S. The output of the low pass filter 75 is rectified negatively in rectifier 77 while the output of the high pass filter is rectified positively in rectifier 78. The outputs of the rectifiers are added and the sum controls a relay 79.

When the instantaneous speech input is voiced, the low frequency filter 75 will have a much greater output than the high frequency filter 76. Since the former is rectified to give a negative voltage and the latter to give a positive voltage, the sum of the outputs of the rectifiers 77 and 78 will be a large negative value which may be used to bias a relay control tube to cut-off and thus open the relay 79. On the other hand, when the sound is unvoiced, the sum of the two rectified voltages will be a large positive voltage which may be used to drive a relay control tube to saturate thus closing the relay 79.

However, there are some marginal sounds for which the output of the two filters will be approximately equal. The bias voltage fed to the grid of a relay control tube would then fluctuate about some intermediate value causing the relay 79 to chatter between its open and closed positions. In order to mitigate this undesirable action, the audio voltage is first fed to a volume compressor 74. The volume compressor 74 is similar to a limiter circuit except that it has a large time constant so that the out-
put of the compressor 74 is still sinusoidal for a sine wave input, the output being smaller multiple of the input for large inputs than for small inputs.

Thus, when there is a large signal input, the volume compressor 74 does not affect the operation of relay 79 because it is immaterial whether the relay is energized very strongly or just energized enough, as far as small signals are concerned, the volume compressor 74 reduces the amplitude of the fluctuations thereby reducing the tendency of the relay 79 to chatter.

The chattering of relay 79 can be further reduced by means of a property common to all relays, that is the current which just makes a de-energized relay to energize is not equal to the current which just causes the energized relay to de-energize. By proper design of the relay armature and contacts, this difference can be accentuated so that a considerable current change is required to first open and then close the relay or vice versa, although the change needed to do only one of these functions may be quite small.

Referring to Fig. 3, wherein this relay effect is graphically illustrated, the x axis representing the magnitude of the control bias and the y axis representing relay closed position above the x axis and relay open position below the x axis. For a control bias A, the relay will always be open and this condition will continue as the bias is made less negative until point B is reached. For a bias between points B and C, the relay will still stay open if it is originally open, that is for the region BC was approached from the left. At the bias C the relay closes and will stay closed for all bias voltages to the right of point C. If the bias voltage is now made less positive, the relay will stay closed until the bias B is again reached. Thus, in order to make the relay operate through a complete cycle, it will require a voltage change of a magnitude BC which may be made large to decrease the chattering of the relay.

Another type of voiced-unvoiced detector which may be used with the communication system of this invention is shown in Fig. 4, wherein the instantaneous speech input is coupled to an amplifier 80 whose output is fed to a non-linear device 81. The instantaneous audio energy is purposely distorted by some non-linear circuit 81, such as a crystal, in order to emphasize the distinction between voiced and unvoiced sound characteristics. Voiced sounds being periodic will have a definite fundamental frequency generally above 80 cycles per second. Distortion of a sound introduces a D-C component along with components at multiples of the fundamental frequency, but there will be no energy at a frequency between the D-C component and the fundamental frequency. Unvoiced sounds being aperiodic when distorted will have a frequency spectrum which is continuous rather than a collection of discrete frequencies as will a distorted voiced sound. Distortion of the unvoiced sound introduces both a D-C component and continuous frequency values from D-C up to high frequencies.

The output of the distortion circuit 81 is fed to a band pass filter 82 which passes the energy contained between 10 and 76 C. P. S. When an unvoiced sound is distorted, energy will be present in this band of frequencies, but a distorted voiced sound, generally having a fundamental frequency above 80 C. P. S., will not have any energy present in this filtered band. This energy output or lack of output may be used to control the presence or absence of a steady tone from a tone generator 83. The tone generator 83 output is used to control the synchronous relay of the communication system shown in Fig. 1.

Still another type of detector applying the distinctive characteristics of voiced and unvoiced sounds to operate the synchronous relay of the communication system shown in Fig. 1 is illustrated in Fig. 5. It is obvious that the voiced sounds having the majority of their significant energy in the lower frequency band will have a greater number of zeros than unvoiced sounds which lie in the upper portion of the speech frequency spectrum. The instantaneous speech output of the speech amplifier 84 which continuously adds a 3 kc. signal of constant amplitude obtained from the microphone 85 to the speech signal. The gain control of the mixer-amplifier 84 is so adjusted that a minimum amplitude signal required to operate the synchronous relay is large enough to reduce the number of zeros considerably.

The output of the mixer-amplifier 84 is limited in circuit 86 until the output of a differentiation circuit 87 will produce only one pulse per second. Each of the positive output pulses from the differentiation circuit 87 fires a gas triode 88 causing a condenser 89, from its plate to ground, to discharge into a second larger condenser 92 in the cathode circuit of the gas triode 88. The time constant of the condenser 89 between plate to ground and the plate load resistor 91 through which it is recharged is small compared to the maximum rate at which zeros may occur, so that the charge added to the condenser 92 in the cathode per pulse is independent of the spacing of the zeros of the speech waveform. The average voltage across the cathode condenser 92 is more negative than the number of zeros per second, and if the average voltage output is high enough when clipped in diode 92, it will bias the grid voltage of a relay control tube 93 to the point where it will close the synchronous relay. This pull in voltage is controlled by the adjustable bias resistor 94 in the grid circuit of the relay control tube. The number of zeros per second required to obtain this voltage is controlled by the rheostat 95 of the cathode circuit in the gas tube 88. The diode clipper 92 in the grid circuit of the relay control tube 93 places an upper limit on the voltage developed by the zero counter-circuit. The purpose of this diode clipper 92 is to minimize the time required for the relay to de-energize when the input comprises a voiced sound following an unvoiced sound or no speech sound input.

Referring to Fig. 6, an improved version of the voiced-unvoiced detector of Fig. 5 is shown, wherein the A-C. output of the zero counter is used in conjunction with the D-C. output to control the bias of a relay control tube 93. The detector of Fig. 5 obtains a D-C. voltage output from the zero counter proportional to the number of zeros per unit time in the sound input. While the majority of speech sounds are characterized either voiced or unvoiced with a low and a high number of zeros, respectively, there are some speech sounds which are intermediate. These intermediate sounds of the zero type detector of Fig. 5 do not categorize well. Interpreting them sometimes as voiced and sometimes as unvoiced sounds causes relay chattering. The detector of Fig. 6 reduces this ambiguity. Here the D-C. output of the zero counter is only part of the measure of whether a sound is voiced or unvoiced. The A-C. output is utilized by passing it through a low pass filter 96 and then a high pass filter 97, which in conjunction constitute a band pass filter from approximately 80 to 180 cycles. The filtered output is coupled to a phase splitter 98 giving two push-pull outputs which are fed to a full wave rectifier 99, the output of which is a negative D-C. voltage proportional to the A-C. voltage amplitude output of the zero counter. Push-pull output is utilized because the ripple components in the rectifier 99 output are most easily filtered. The D-C. negative voltage output of the rectifier 99 is then added to the original loudness voltage at the zero counter output and the sum is used to control a steady tone from the grid of a relay control tube. This sum is a more accurate indication of whether a sound is voiced or unvoiced.

While we have described above the principles of our invention in connection with specific apparatus, it is to be clearly understood that this description is made only by way of example and not as a limitation to the scope
of our invention as set forth in the objects thereof and in the accompanying claims.

We claim:

1. A communication system for the transmission, within a narrow frequency band, of speech signals having a wide frequency bandwidth comprising means to detect in said speech signals the voiced and unvoiced sounds, means to generate an identifying signal of predetermined regular characteristics different for each of said sounds responsive to the output of said detecting means, means to separate the speech signals according to their frequencies into narrow sub-bands of said frequencies different for each of said sounds, means for shifting at least one of said sub-bands of frequencies so that said sub-bands to the same portion of the frequency spectrum to form a narrow audio frequency band signal retaining the speech character of each of said sub-bands, means for transmitting said narrow audio frequency band signals, without destroying their speech character, together with said identifying signals, to a receiver to detect said transmitted signals, and means for transposing said received narrow sub-bands of frequencies to their original frequency allocation in the speech signal spectrum responsive to said identifying signals.

2. A communication system for the transmission, within a narrow frequency band, of speech signals having a wide frequency bandwidth comprising means for detecting the voiced and unvoiced component sounds of said speech signals, switching means responsive to the output of said detecting means, means to form a different narrow sub-band of frequencies from each of said speech signal components responsive to said switching means, means to generate an identifying signal responsive to the output of said detector means, means for relaying said sub-bands to the same portion of the frequency spectrum to form a narrow frequency band signal, means for transmitting said narrow frequency band signals and said identifying signal, a receiver to detect said transmitted signals, and means for transposing said received narrow sub-bands of frequency to their original frequency allocation in the speech signal spectrum responsive to said identifying signals.

3. A system according to claim 2, wherein said means for forming a narrow sub-band of frequencies from each of said speech signal components further includes means to delay the input speech signal so that forming means to allow time for the operation of said switching means responsive to said detector.

4. A system according to claim 2, wherein said means for detecting the voiced and unvoiced components of said speech signals comprises filtering means for the low frequency energy of said input speech signals, means to rectify said filtered low frequency energy, filtering means for said high frequency energy, means to rectify said filtered high frequency to a polarity opposite of said rectified low frequency energy, means to add said rectified energy whereby the polarity of the sum of said rectified energy is indicative of the input speech signal component, and means to reduce the amplitude of said input signal to the two said filtering means inversely to the original amplitude of said input signal.

5. A system according to claim 2, wherein said means for forming the voiced and unvoiced components of said speech signal comprises means to distort said speech signal input and filtering means to pass a predetermined frequency band less than the fundamental frequency of a voiced sound whereby an output from said filtering means will be indicative of an unvoiced sound component input while lack of an output from said filtering means will be indicative of a voiced sound component input.

6. A communication system for the transmission, within a narrow frequency band, of speech signals having a wide frequency bandwidth comprising means for detecting the voiced and unvoiced components of said speech signals including means to limit the input signal, means to differentiate the output of said limiter means, means to generate a signal responsive to the number of times said differentiator output waveform crosses the zero amplitude axis whereby the amplitude of said generated signal should be indicative of the characteristics of the voice signal component input, means to generate a relatively high frequency signal, means to mix said generated signal and said speech input, and means to couple the output of said mixing means to said limiting means, switching means responsive to the output of said detecting means, means to form a different narrow sub-band of frequencies from each of said speech signal components responsive to said switching means, means to generate an identifying signal responsive to the output of said detector means, means for relaying said sub-bands to the same portion of the frequency spectrum to form a narrow frequency band signal, means for transmitting said narrow frequency band signals and said identifying signal, a receiver to detect said transmitted signals, and means for transposing said received narrow sub-bands of frequency to their original frequency allocation in the speech signal spectrum responsive to said identifying signals.

7. A communication system for the transmission, within a relatively narrow frequency band, of a plurality of speech signals each having a wide frequency bandwidth comprising means to separate each of said speech signals into its voiced and unvoiced sound components, filtering means to rectify said separated sounds, filtering means to rectify said separated sounds to a polarity opposite of said rectified low frequency energy, means to add said rectified energy whereby the polarity of the sum of said rectified energy is indicative of the input speech signal component, and means to reduce the amplitude of said input signal to the two said filtering means inversely to the original amplitude of said input signal.

8. A communication system for transmission, within a relatively narrow frequency band, of a plurality of speech signals each having a wide frequency bandwidth comprising means to separate each of said speech signals into its voiced and unvoiced sound components, means to generate a basic modulating frequency, means to distort said basic modulating frequency to provide a plurality of modulating signals, and means to modulate at least one of said modulated signals to transmit said separate modulated frequencies to said transmitting means.

9. A system for transmitting, within a narrow frequency band, speech signals having a wide frequency bandwidth comprising means to detect said speech signals the voiced and unvoiced sounds, means to transposing said transmitted components of each speech signal, and means responsive to said detected identifying signals to restore said transmitted components to their original position in the frequency spectrum.
signals according to their frequencies into narrow sub-bands of said frequencies different for each of said sounds, means for shifting at least one of said sub-bands of frequencies to bring said sub-bands to the same portion of the frequency spectrum to form a narrow audio frequency band signal retaining the speech character of each of said sub-bands, and means for transmitting said narrow audio frequency band signals, without destroying their speech character, together with said identifying signals.

10. A system for transmitting within a relatively narrow frequency band a plurality of speech signals each having a wide frequency bandwidth comprising means to separate each of said speech signals into its voiced and unvoiced sound components, filtering means responsive to said separating means to separate the significant energy frequencies of each of said components, means to generate a basic modulating frequency, means to distort said basic modulating frequency to provide a plurality of modulating signals, means to modulate at least certain of said significant energy frequencies of each of said components to transpose said significant energy frequencies of each speech signal to the same portion of the frequency spectrum, means responsive to the output of said separating means to generate a plurality of identifying signals, and means to transmit said basic modulating frequency and said transposed components and said identifying signals.

11. A device to separate the high frequency and low frequency components of an input signal comprising means to limit the input signal energy, means to differentiate said limited output, means to generate a signal having an amplitude proportional to the number of times said differentiator output waveform crosses the zero amplitude axis, and switching means responsive to said generated signal for effecting separation of said high and low frequency.

12. A device according to claim 11, which further includes means to generate a constant amplitude signal and means to mix said generated high frequency signal and said input signal whereby the distinct characteristics of the high frequency and low frequency components of said input signal are emphasized.

13. A device to separate the high frequency and low frequency components of an input signal comprising means to limit the input signal energy, means to differentiate said limited output, means to generate a signal having an amplitude proportional to the number of times said differentiator output waveform crosses the zero amplitude axis, including a gas triode space discharge device operatively responsive to the output of said differentiating means and a condenser coupled to the output of said gas triode whereby the average voltage across said condenser is proportional to the output of said differentiating circuits, and switching means responsive to said generated signal for effecting separation of said high and low frequency.

14. A device according to claim 13, which further includes a band pass filter for the alternating voltage output of said gas triode, means to split the phase of said filtered alternating voltage, means to rectify output of said phase splitter, and means to add rectified output to said voltage across said condenser.

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