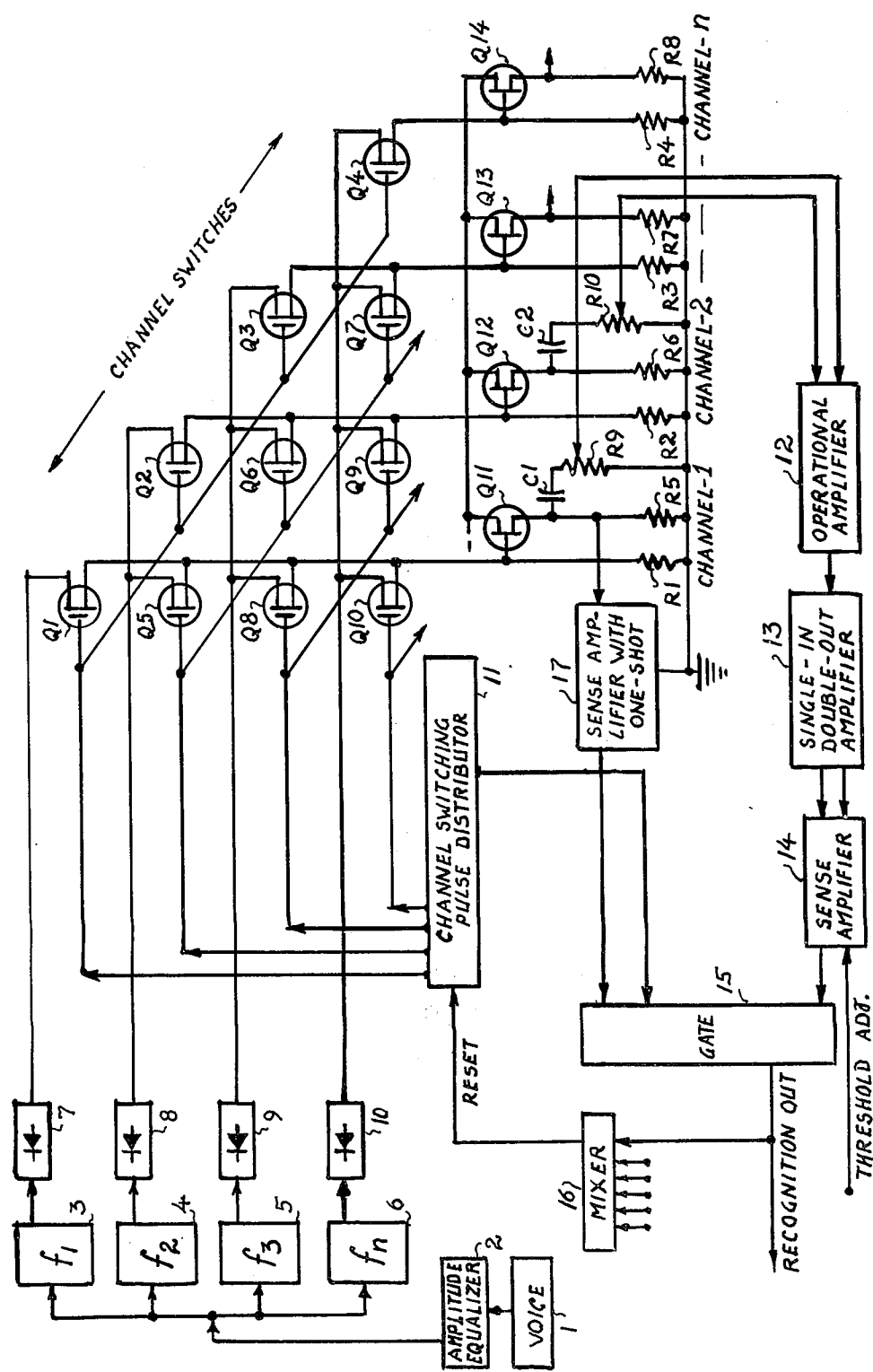




PHONETIC SOUND RECOGNIZER

This invention relates to speech sound wave analysis, and more particularly to the provision of control signals for indicating the true or false nature of the final signal derived as a representation of the desired phonetic information.

In my previous patent disclosures, for example, U.S. Pat. No. 3,432,617 issued Mar. 11, 1969, I have described that the phonetic information, as spoken by different speakers, is represented by the combination of both the frequency and amplitude ratios between the resonances of a select group of resonances in the complex sound wave. In that patent disclosure, I had described a null indicating circuit with prefixed gain adjustments, so that when two signals having predetermined amplitude ratio between the two arrive simultaneously at the inputs of this circuit within a given analytical time period, a null output will be obtained as an indication of the phonetic information desired. In such an arrangement, however, said null output will also be obtained during said time period when both of the input signals to the circuit are absent, resulting false indication of the desired phonetic information. In order to avoid such false indication, a sensing means is included in the complete phonetic information analyzing system, as disclosed herein, so as to allow production of the final information representing signal only when said input signals are present. This will be described in more detail in the following specification, when read in connection with the accompanying drawing.

In order to understand the purpose and usefulness of the present invention in a system of speech sound recognition, the drawing consists of a complete system of phonetic sound recognition, wherein, the voice sound wave in block 1 is first amplitude equalized in block 2, and applied to a set of band-pass filters in blocks 3 through 6. These filters are used to sub-divide the frequency spectrum of speech sound waves, and the outputs of these band-pass filters are detected in blocks 7 through 10, respectively. The outputs of these detectors are then analyzed for deriving the desired information as representation of a phonetic sound in the spoken sound wave.

Due to the enormous amount of spectral variations that occur in normal speech sound waves, I have described in my patent issue No. 3,622,706 issued Nov. 23, 1971, a system of spectral normalization by way numerical identity conversion in a special arrangement of numerically identified switchable channels. I have also described a simplified and more practical arrangement of the switchable channels in my patent application Ser. No. 368,265 filed July 8, 1973, now U.S. Pat. No. 3,864,518, issued Feb. 4, 1975, which is included in the present disclosure representing the channels 1 through *n*, in the accompanying drawing. These channels are numerically identified, and each one consists of a number of channel-switching AND-gates depending upon the numerical location in the numerical sequence, for example, the channel-1 consists of the largest number of AND-gates; the channel-2 consists of one less than that number; the channel-3 consists of one less than the number of AND-gates in the preceding channel; and so on. These AND-gates are represented by the channel-switching transistors Q1 through Q10, which admit input signal to output only when simultaneous input switching signal is present. Thus, the drawing shows that the outputs of blocks 7 through 10 are applied to

one of the input electrodes of Q1, Q5, Q8, Q10, respectively, the outputs of which are connected in parallel to the input of output-transistor Q11, which allows admittance of any one of the output signals of detector blocks 7 through 10 to the input of output-transistor Q11 by applying switching pulse to the other input electrodes of the gate-transistors by the pulse distributor in block 11. Besides such multiple switching of selected input (detected) signal to the channels, for example, channel-1, the switching input electrodes of these AND-gates are connected in parallel in an order, such that when the second (numerically identified) input signal is switched to the input of the first channel, the third input signal is switched to the input of the second channel, as represented by the transistor Q12, and by the parallel connections of the signal-input electrodes of the AND-gates, for example, Q5-Q2; Q8-Q6-Q3; and Q10-Q9-Q7-Q4. Such an arrangement facilitates shifting of the input signal to the channels in such numerical order that, the signal having the lowest numerical identity at the outputs of the detectors in blocks 7 through 10 is shifted to the first channel, with proportional numerical shifting of the other signals to respective channels. This channel-identity numerical conversion is important for final analysis of the information to be derived from the detected signals. For example, if the numerical ratios between the numerically identified group of detected signals represent the information desired, this group of signal may arrive at the detectors such that the signal having the lowest numerical identity may arrive at any one of the detector blocks 7 through 10, because of spectral variation in the voice sound spectrum from block 1. Whereas, by channel identity shifting, such that the signal having the lowest numerical identity is shifted to the output of the first channel, the selection of these signals from standard numerical locations from the channel outputs becomes much simpler for final analysis by matching with standard parameters. Such simple numerical channel switching, however, also requires sub-division of resonances in the sound spectrum by the band-pass filters in an order of harmonically increasing series of frequencies, and said series comprising a plurality of sub-series, all sub-series having the same number frequencies such that each frequency in a sub-series is harmonically related to the same-placed frequency in the other sub-series. These band-pass filters are then numbered sequentially for numerical identity in the same numerical sequence of the channels, as shown in the drawing, so as to make the channel identity shifting of the numerically identified detected signals without destroying their original numerical ratio identities. This has been more fully explained and illustrated in my reference patent application, now patent No. 3,864,518, issued Feb. 4, 1975. By such spectral normalization, in the mode of channel identity shifting, any group of signals that may be selected from the outputs of the band-pass filters in blocks 3 to 6, will be shifted to a standard numerical location at the outputs of the channels, and thereby, selection of the important output signals from the output-transistors Q11 to Q14 will be standardized for simple analysis. This is simply accomplished by the pulse distributor in block 11, which activates the switching inputs the said AND-gates sequentially until one of the switching operations position a desired group of signals from the outputs of the detectors 7-10 in the correct numerical location at the outputs of the channels.

After spectral normalization has been established, the final step of information analysis is the measurement of the amplitude ratios between the selected signals. This is done, for example, by measuring the signal amplitude ratio between the output signals of the output-transistors Q11 and Q12, from across output resistors R9 and R10. With the assumption that the signal amplitude ratio is consistent for a particular information, for example phonetic sound, then we may make fixed tap-adjustments across these resistors, so that the signal amplitudes across these resistors will be equalized whenever this particular information arrives. Recognition of the equalized state of two signals becomes simple by null-signal identification, as in the following:

With the above given explanation, assume now that the signals at the taps across output resistors R9 and R10 are of equal amplitudes as representation of a particular information, during a pulse period from the distributor in block 11. These equalized signals are applied to the inputs of an operational amplifier in block 12, which by virtue of its invert and non-invert inputs, the net output will be zero (null) signal, or at least below a threshold minimum amplitude level; this zero output representing the desired information. The output of block 12 is applied to an amplifier in block 13, which converts the single-ended input into a push-pull output. This push-pull output is then further applied to a sense amplifier in block 14, which produces an output signal of prefixed polarity, regardless of the polarity of the arriving signal at its push-pull input. The purpose of such an arrangement is that, when the input signals of block 12 are of different amplitudes, representing false information, the single ended output of block 12 may be in either positive or negative polarity, depending on which of the two input signals is larger than the other. Since commercially available sense amplifier integrated circuits are usually provided with double-ended inputs, the amplifier in block 13 is used for the required signal polarity conversion. The sense amplifier in block 14 is usually provided with a threshold level adjustment, and it can be manually adjusted to a fixed level, so that the sense amplifier in block 14 will produce an output signal only when its input signals are above a fixed threshold amplitude level.

With the functional operations of the amplifiers in blocks 12, 13 and 14, as described above, the output of sense amplifier in block 14 will be in quiescent state when the input signals of block 12 are of equal amplitudes, and in ON-state when the input signals of block 12 are in different amplitudes. Assuming thus, that when the arriving signals at the inputs of operational amplifier in block 12 are of equal amplitudes, and the output of sense amplifier in block 14 is at 0 level during a distribution pulse period, both the distribution pulse and the output sense amplifier are applied at 0 levels to the AND-gate in block 15, which in turn produces an output pulse at 1 level. This output pulse of 1 level may then represent the information desired, and may be utilized in any desired mode, for example, for the operation of a key of a typewriter. In the case that the input signals of the amplifier in block 12 are of different amplitudes during the pulse period of the distributor in block 11, then the output of sense amplifier in block 14 will be in ON-state (1 level), and the 0 level distribution pulse applied to the AND-gate in block 15 will not operate, thus avoiding false signal output from the gate circuit. The signal amplitude balancing is

shown between only two signals, but of course, more than two signals can be balanced within the pulse period of the distributor, such as described and illustrated in my related patent application Ser. No. 368,264 filed June 8, 1973 now U.S. Pat. No. 3,864,518, issued Feb. 4, 1975, the illustration of which is only exemplary, and commercially available integrated devices suitable for the purpose may be used.

With the above given explanation, it will be noted that the presence of the desired information is determined by the OFF-state of the sense amplifier in block 14. This OFF-state will also prevail when the input signals to the operational amplifier in block 12 are both absent during a distribution pulse period. In order to avoid such false indication of information, each pair of signals at the channel outputs is first checked to make sure that one or both of the signals are present. This is easily accomplished by applying the output of the first channel (from the source circuit resistor R5) to one of the inputs of the inputs of the sense amplifier in block 17. With the presence of signal at its input, the sense amplifier 17 will produce an output pulse by the one-shot that it may contain, as available commercially. The operating time of this one-shot may be adjusted to be equal to the pulse time period of the distribution pulse from block 11, so that during a distribution pulse the sense amplifier will simultaneously produce a similar output pulse at 0 level when an input signal is present at the input of channel-1. This pulse from the output of sense amplifier in block 17 is simultaneously applied to the three-input gate in block 15, so that only when the three inputs of the gate are simultaneously at 0 level will cause operation of the gate, as a representation of the information desired, thus making sure that false indication of information is not performed. The distributor in block 11 may be free running ring distributor, or it may be reset after an information signals has been derived, by applying the output of gate 15 to the mixer in block 16, and further applying it to the reset input of the distributor, as shown.

Having described the preferred embodiments of the invention disclosed herein, and in view of the broad scope of uses it may embrace in practice, it is obvious to the skilled in the art that it may be considered as exemplary, and therefore, various modifications, adaptations, and substitutions of parts may be made without departing from the true spirit and scope of the invention.

What I claim, is:

1. A signal matching system for deriving an information-representing signal from the signal-amplitude-ratio between first and second signals when said ratio matches with an arrangement of prefixed signal-amplitude-ratio adjustments in first and second channels during a predetermined time period in which a control signal is produced, said system comprising means for producing said control signal; first and second channels; coupling means from said first and second signals to said first and second channels, said channels having prefixed signal-amplitude-gain adjustments, so that the signal-amplitudes of said first and second signals are equalized at the outputs of said first and second channels in the form of a null signal only when the signal-amplitude-ratio between said first and second signals matches with said prefixed adjustments; a signal sensing means for deriving a sensing signal from said first and second signals; and means for deriving a dis-

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crete signal representing said information from the combination of said null-signal, said control-signal, and said sensing-signal, last said signal making sure that said null-signal is derived only from the presence of said first and second signals, and not from their absence in said channels.

2. The system as set forth in claim 1, wherein said first and second channels consist of an operational amplifier for producing said null-signal at its output.

3. The system as set forth in claim 1, wherein is in-

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cluded means for translating said null-signal, said control-signal, and said sensing signal into similar-polarity signals; a multi-input gate; and means for applying said similar-polarity signals to the inputs said gate for obtaining said discrete signal.

4. The system as set forth in claim 1, wherein is included coupling means from said discrete signal to said means for producing said control-signal for resetting them to normal states after said discrete signal is used.

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