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METHOD FOR QUICK GLITCH FILTERING IN SYSTEMS FOR AUTOMATED SPEECH RECOGNITION

There had been suggested the algorithm and the device for quick signals glitch filtering in speech signals based upon the modified method of smoothing by moving average.

***Key words**: filtering, disturbing signal, approximation, gain of filter, computational complexity, optimized frequency response.*

Problem statement. Research, conducted by authors in [1,2] showed that under specific conditions noise filtering in speech signal may cause the significant improvement in accuracy of signal recognition. However, the known methods of filtering require significant calculating costs, which does not allow to realize recognition in on-line mode. In view of this, the search for filtering methods which require minimum calculating costs of computer or those, to be realized on the base of simple digital devices with quick response, remains the topical issue. Concerning the speech signal, this task may be solved from the point of building the smoothing filter.

Analysis of researches and publication on this mater and setting objectives. The knowing classic methods for smoothing which find signal approaches by set functions [3], may not be applied due to some reasons . First, the type of the curve with which the approximation must be done, is unknown. Second, such smoothing requires the availability of information on all counting of decrease signal and in on-line mode is impossible. Under such conditions building of smoothing filter, approximated by the polynomial of the first degree, that is, using the method of moving average, is reasonable. The paper suggests an algorithm and the device for quick glitch filtering in speech signals, based on the modified method for smoothing by moving average.

Mathematical substantiation of filtering method. The number of "moving" group is determined proceeding from requirements for smoothing accuracy and the minimum of deviations quadrant sum of calculations of the real signal from the smoothed one as the criteria of smoothing optimality:

$$\sum_{i=1}^{m}(y_i - y_i^*)^2 = \min , \tag{1}$$

where y_i – the current value of the signal on the filter input in the i-th time moment, y_i^* – the smoothed current signal counting, m – number of countings.

For smoothing by "moving" group from three points from condition (1) we receive the following expression for counting the smoother signal.

$$y_i^* = \frac{1}{3}y_{i-1} + \frac{1}{3}y_i + \frac{1}{3}y_{i+1}, \tag{2}$$

where y_i^* –smoothed current signal counting and y_{i-1}, y_i, y_{i+1} – three consequent in time signals countings on the filter input.

The realization of smoothing algorythm on this formula is not optimal from the point of view of calculating costs since the operations of deviding require most of time and hardware costs. The frequency characteristics of such smoothing filter is not optimized, since it suppress the useful components, which are below the frequency of signal digitization. To demonstrate this we find

z- transformation of the expression (2) and the function of the filter:

$$Y(z)\cdot z^{-1} = \frac{1}{3}X(z)\cdot(z^{-2} + z^{-1} + z\ ), \tag{3}$$

$$W^*(z) = \frac{Y(z)}{X(z)} = \frac{1}{3}(z^{-1} + z + 1). \quad (4)$$

The frequency characteristics of such filter looks like:

$$H^*(j\omega) = \frac{1}{3}(1 + e^{-i\omega t} + e^{i\omega t}) = \frac{1}{3}(1 + 2\cos \ \omega t). \quad (5)$$

Fig. 1, on which curve 1 demonstrates the module of frequency characteristics of this filter, shows, that in frequency range $[\frac{2\pi}{3} \cdot\cdot \pi]$ there is the suppression of useful components of speech signals. In general case the frequency characteristics of the device, which realizes smoothing by "moving" group from three points, looks like:

$$H(j\omega) = (a_0 + 2a_1 \cos\omega t). \quad (6)$$

Let's assume the requirement according with which the module of transmissive function equals zero on the zero frequency, and on the highest frequency of the signal-0:

$$\begin{cases} a_0 + 2a_1 \cos(\omega T) = 1, \ если \ \ f_c = 0; \\ a_0 + 2a_1 \cos(\omega T) = 0, \ если \ \ f_c = \frac{1}{2} f_d = \frac{1}{2T}, \end{cases} \quad (7)$$

where a_0, a_1 – factors of smoothing filter, ω – angular frequency, T – period of digitization, f_c – signal frequency f_d – digitization frequency .

From system (7) we find that $a_0 = \frac{1}{2}$, $a_1 = \frac{1}{4}$. Substituting these values in (6), we receive the expression of the transmissive function of filter:

$$H^*(j\omega) = \frac{1}{2} + 2 \cdot \frac{1}{4}\cos\omega t. \quad (8)$$

Module of frequency characteristics of the obtained filter is presented by the curve 2 in Fig. 1.

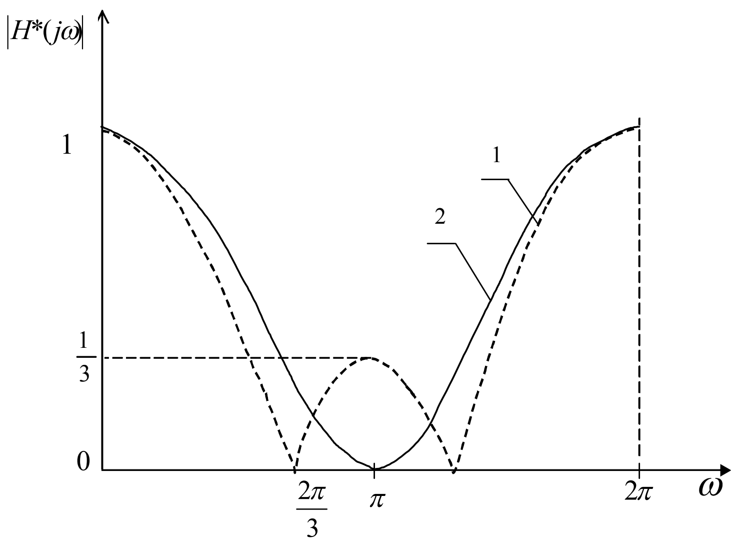

Fig.1. Frequency characteristics of the known (1) and suggested (2) filters

The transmissive function of filter, corresponding to this algorithm of signal processing is:

$$W(z) = \frac{1}{4}z^{-1} + \frac{1}{2}z^0 + \frac{1}{4}z^{+1}. \quad (9)$$

In time sphere, the filter may be represented as convolution of the sequential signal, counts with factors of digital filter:

$$y_i^* = (y_i + \frac{y_{i-1} + y_{i+1}}{2})/2 = \frac{1}{4} y_{i-1} + \frac{1}{2} y_i + \frac{1}{4} y_{i+1}. \quad (10)$$

Quantitative decrease in smoothing efficiency, using the suggested methods in comparison with classical one may be determined, finding the influence of digital filter on the level of noise in the signal. Let $y(k) + \varepsilon(k)$ – the additive mixture of the signal $y(k)$ and noise $\varepsilon(k)$, and mathematical expectation of the noise $M[\varepsilon(k)] = 0$, where mathematical expectation M shall be determined as the ensemble – averaged of the noise signals. Let's also assume that for factual mixture values $y(k) + \varepsilon(k)$ the noise countings are non-correlated, that is:

$$M[\varepsilon(k_1), \varepsilon(k_2)] = \begin{cases} \sigma_\varepsilon^2, k_1 = k_2, \\ 0, k_1 \neq k_2 \end{cases}, \quad (11)$$

where σ_ε – noise dispersion.

The output signal of non-recursive filter is determined according to the formula:

$$y^*(k) = \sum_{i=-N_i}^{N_i} a_i [y(k-i) + \varepsilon(k-i)],$$

где a_i – filter factors.

Since the averaging is only applied to $\varepsilon(k)$, the most probable signal value on filter output equals:

$$M[y^*(k)] = \sum_{i=-N_i}^{N_i} a_i \{y(k-i) + M[\varepsilon(k-i)]\} = \sum_{i=-N_i}^{N_i} a_i y(k-i).$$

Let us calculate the variance of the input signal:

$$M\{[\sum_{i=-N_i}^{N_i} a_i \{y(k-i) + \varepsilon(k-i)) - M(y^*(k))]^2\} = M[\sum_{i=-N_i}^{N_i} a_i \varepsilon(k-i)]^2 =$$

$$= M\{[a_i \varepsilon(k-i)][\sum_{i=-N_i}^{N_i} a_i \varepsilon(k-i)]\}.$$

Multiplying the expression in brackets and considering (11), we get:

$$\sum_{i=-N_i}^{N_i} a_i^2 M(\varepsilon^2) = \sum_{i=-N_i}^{N_i} a_i^2 \sigma_\varepsilon{}^2 = \sigma_\varepsilon{}^2 \sum_{i=-N_i}^{N_i} a_i^2. \quad (12)$$

From (12) it is seen, that the suppression of noise by filter is determined by the quadrant sum of its factors, therefore the efficiency decrease in filtering by filter (10) in comparison with filter (2) shall be defined by the expression:

$$E_f = \frac{\sigma_\varepsilon{}^2 \sum_{i=-N_i}^{N_i} a_{ip}^2}{\sigma_\varepsilon{}^2 \sum_{i=-N_i}^{N_i} a_{i0}^2}, \quad (13)$$

where a_{ip} – factors of this filter, a_{io} – factors of optimum filter.

Substituting in expression (13) the values of factors a_{ip} –the suggested digital filter and a_{io} – classical filter, we receive the quantitative value of decrease in smoothing efficiency:

$$E_f = \frac{(1/2)^2 + 2\times(1/4)^2}{3\times(1/3)^2} = 1{,}68\,.$$

Filter (10) realizes the calculation of factors by shifting the sequential signal countings and this excludes the operations of multiplying and deviding, necessary for filter(2). Simple calculations show that the response of filtering increases by an order or higher. This, the suggested filter has the response which is by an order higher than that of the optimal filter with insignificant decrease in filtering efficiency. Such non-recursive filter may be realized in a hardware as a chain of buffer registers, which outputs are switched to the input of the summator (Fig. 2).

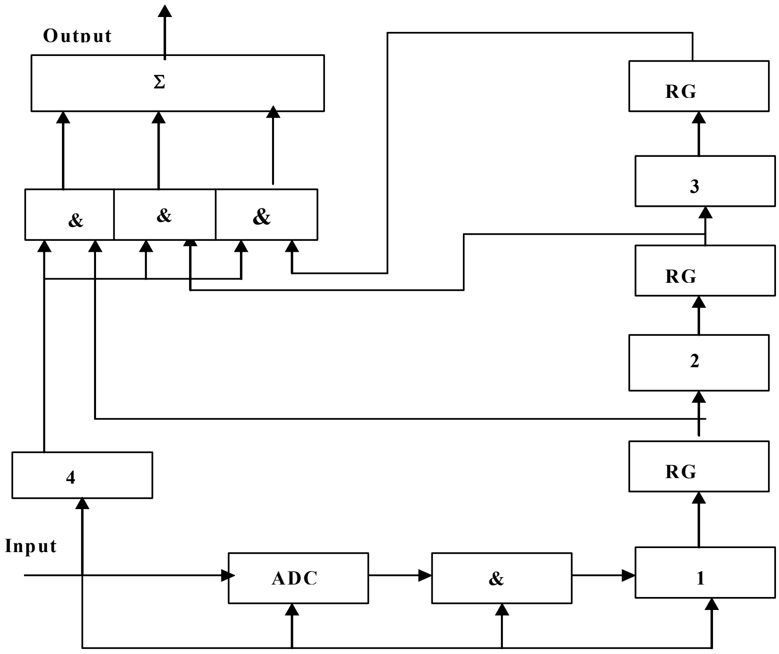

Fig. 2. Structural diagram of the device for filtration by the suggested method

Fig.2 uses the following designations : ADC – analogue-to-digital converter, RG – registers, 1 – chain of shifting the binary worol by two digits to the right , 2 – chain of shifting by one digit to the left, 3 – chain of shifting by one digit to the right, 4 – block for information output resolution, Σ – binary summator, & – coujunctors.

The device operates as follows: Signal values, measured in the discrete time periods, are transferred to the digital binary code (binary word) by the ADC). This very time periods ADC creates tackt pulses for synchronization of the whole device. During the transfer of binary words, defining the signal counting, they are shifted from one register to the other by chains so that the content of the first and the third register equals $y_{i-1}/4$ and $y_{i+1}/4$ correspondingly, and the content of the second register equals $y_i/2$.

The smoothed signal counting shall be received by summing the content of three registers on each fact pulse. Block for information output resolution 4 in the initial time moment prohibits reading the content of registers up to the moment of filling all three registers. This device executes the operation of deviding simultaneously with shifting codes of all the components during the transfer of signal countings from register to the register . Since there is no necessity in operation of deviding, the arythmetic device is significantly simplified in comparison with that for executing calculations on the smoothing formula (2) as it does not need the device for executing deviding to be a part of it.

The gain in response of the suggested smoothing filter in comparison with filter (2) may be calculated on the formula:

$$e_w = \frac{k_d(m_d - n_d)T}{T}, \tag{14}$$

where k_d – number of microoperations in microprogram,which realizes the deviding algorythm; n_d – word length of the divisor; T – duration of the beat for execution of microoperation.

We accept $m_d = 8$, $k_d = 6$, $n_d = 2$ (word length of the digit 3), receive e_w=6(8-2)=36.

Basing on the theory of function approximation [4] and theory of digital filters [5], allows to demonstrate that the connection of such elementary filters enables to build a quick responding filter with set characteristics . The proof for this statement is beyond this work.

Conclusions. This paper suggests the algorithm and the device for quick filtering signals noise in speech signals, based on the modified smoothing method by sliding averages. The suggested filter realizes the calculation of factors by shifting the sequence signal countings and in such a way avoids the operations of multiplying and dividing, which are necessary for classical filter. It had been proved mathematically, that the smoothing filter, received with the consideration of the optimization of frequency characteristics has the response, which is higher by an order in comparison with the known ones, which allows to apply it for filtering the speech signal in the on-line mode . The insignificant decrease in filtering efficiency is observed.

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