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THE METHOD OF A FAST FILTERING FOR NOISE REDUCTION IN AUTOMATIC SPEECH RECOGNITION SYSTEMS

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The analysis given in the previous part has shown that the filtration of noise in a speech can result to considerable increasing of the recognition accuracy under definite conditions. According to this there is a problem of looking for the filtration methods which require computers with minimum computing recourses or digital devices constructed on the basis of foolproof with high rate. In the interval of time the filtration can be evaluated by the convolution of consistent signal samples with factors of a digital filter.

In this paper the filter with fast response time and efficiency of filtration close to optimum is reviewed. The algorithm of implementation:

$$y_i^* = (y_i + (y_{i-1} + y_{i+1})/2)/2 = y_{i-1}/4 + y_i/2 + y_{i+1}/4, \qquad (1)$$

where y_i^* is sliding current sample of a signal, and y_{i-1}, y_i, y_{i+1} are three consequent in time samples of an input signal of the filter.

Let's compare response and efficiency of a filtration of the offered filter to the same characteristics of the optimum three-point filter. Here the optimality is evaluated from the point of minimization of root-mean-square deviation of an actual signal from slided signal:

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

The operating algorithm of the optimum filter in this case:

$$y_i^* = \frac{1}{3}y_{i-1} + \frac{1}{3}y_i + \frac{1}{3}y_{i+1}. \qquad (3)$$

The reduction of a noise by the filter is determined by the sum of squares of its factors, therefore the decreasing of a filtration efficiency by the filter (1) in contrast to the filter (5) will be defined by the expression:

$$E_f = \sum_{i=1}^{m}k_{ip} / \sum_{i=1}^{m}k_{io} = ((1/2)^2 + 2\times(1/4)^2)/(3\times(1/3)^2) = 1{,}68, \qquad (4)$$

where k_{ip} – factors of the considered filter, k_{io} – factors of the optimum filter.

The filter (1) realizes calculations of factors by shifting consistent signal samples and because of this eliminates operations of multiplying and divisions what’s indispensable for the filter (3). That’s why the offered filter has response higher on the order that the optimum filter has by a small decrease of filtration efficiency. Such hardware as nonrecursive filter is possible to realize by the way chains of the buffer registers, the outputs of which are connected to the summator input.