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Adjustment of Adaptive Sum Comb Filter for PPG Signals

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***Abstract*—AC component of photoplethysmography signal carries important information for diagnostics. Registered signal may be affected by noises, which are sharing the same bandwidth. Adaptive comb filter is used for the AC component extraction. Due to filter averaging behavior it decreases the signal shape difference between consecutive beats. Comb filter needs to be adjusted for PPG signal. Comb filter new weight values are determined through numerical computation. Experiments with generated photoplethysmographic signals were carried out to compare adjusted and non-adjusted adaptive sum comb filter.**

I. INTRODUCTION

Photoplethysmography (PPG) is optical non-invasive method to measure the blood content changes in observed tissue. For the PPG signal registration, optical radiation from light source is emitted to the observed tissue. Emitted light in tissue is scattered, absorbed, reflected and only small fraction is received by photodetector.

Registered PPG signal consists of DC and more than ten times smaller AC component. AC component is pulsatile and synchronous with heart rate. PPG signal can be used to monitor parameters such as pulse rate, respiratory rate, blood pressure, pulse wave transit time[1] etc. For those parameters PPG signal AC component carries important information.

PPG signals, which are registered while the subject is in motion, e.g. during long term monitoring [2], can be with low signal-to-noise ratio (SNR). The signal needs to be processed to extract the AC component. In previous studies various processing methods have been used [3, 4]. Band-pass FIR filter is one of the simplest method to remove noise. Unfortunately this method is not suitable for the case, where the different noises are sharing the same frequency band with PPG signal.

Adaptive sum comb filter with ECG reference have been proposed in previous studies [5] and used successfully to remove the motion caused noises from the PPG signal. Due to comb filter frequency properties, it extracts the PPG signal harmonical components and suppresses the noises between them. By nature the comb filter averages periods. It has been shown, that it is possible to extract PPG signal shape from noisy signal source. The filter main drawback is in its averaging behavior, which causes the decrease in signal shape difference between beats. In this article the adaptive comb filter adjustment for PPG signal has been performed to decrease the filter effect on signal shape averaging.

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II. METHODS

Sum comb filter is described with next equation [6]:

$$y_k = \frac{x_k + x_{k-D}}{2} \quad , \tag{1}$$

where x_k is input signal, k is sample number, y_k is filter output signal and D is delay of integer number of samples. The filter frequency response exhibits series of regularly spaced band-pass peaks at multiples of the fundamental frequency:

$$f_1 = \frac{f_s}{D} \quad . \tag{2}$$

The frequency response of this filter is:

$$H(j\omega) = \frac{1}{2} \cdot \left(1 + e^{-j\omega D}\right) . \tag{3}$$

The frequency response plot, in case of D=8, is given on Fig. 1(solid line). Based on equation (1) the sum comb filter takes the average of samples x_k and x_{k-D}. Filter delay, D, equals with the period (in samples) of the fundamental frequency. It comes out, that sum comb filter is taking average between two consecutive periods with length of D.

The averaging can be enlarged over the number of r periods. The filter output is calculated then by following equation:

$$y_k = \frac{1}{r} \cdot \left(x_k + x_{k-D} + x_{k-2D} + \ldots + x_{k-(r-1)D}\right) \quad . \tag{4}$$

Respectively the frequency response of this filter is:

$$H(j\omega) = \frac{1}{r} \cdot \left(1 + e^{-j\omega D} + e^{-2j\omega D} + \ldots + e^{-(r-1)j\omega D}\right) \quad . \tag{5}$$

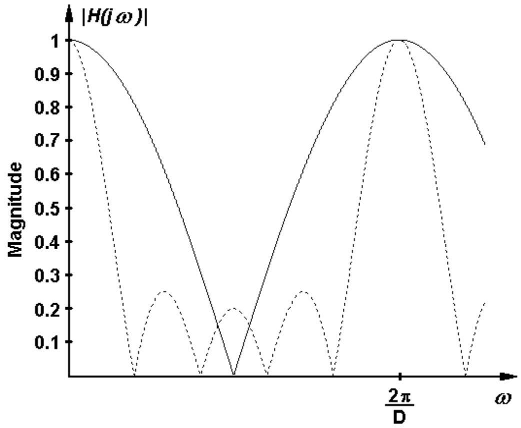

Fig. 1. Sum comb filter frequency responses with D=8. Frequency response is obtained from equation (3) (solid line). Frequency response is obtained from equation (5) in case of r=5 (dashed line).

On Fig. 1(dashed line) is given sum comb filter frequency response plot in case of r=5 and D=8. It is visible, that this filter has better noise suppression in stop-bands than filter, which uses equation (1) algorithm. Filter noise suppression is enhanced in stop-bands by increasing the number of periods, which is used for averaging. First two columns, on Table 1, express the relation between sum comb filter first side lobe attenuation and number of periods, which are used for filter output calculation. It is visible, that by enlarging the r over 4 or 5, it is not giving much advantage in attenuation.

The delay, D, is kept constant during entire filtering process, in case the extracted signal is rigorously periodic. Bio-signals, e.g. ECG and PPG, which are related to heart, are recurring but not periodic. This means that the PPG signal beat-to-beat time is not constant, although the shapes of different beats are similar. In addition for every recurrence new delay value D has to be set.

The sum comb filter is customized for recurring signal. For the current recurrence, which is under process, the filter delay, D, is calculated from the referenced ECG signal. Subsequently, the r consecutive recurrences are rescaled to the length of current recurrence. For the next step, the equation (4) is applied to calculate the filter output within current recurrence. The previously described sequence is applied for every recurrence.

From equation (4) it follows, that every recurrence is taken with the same weight into the comb filter output calculation. In this way the small changes in PPG signal shape, which may be important for the diagnostics, are averaged and may disappear.

To deal with this problem, the filter can be adjusted in the point of view by taking less information from the more elapsed recurrences and more information from the recurrences, which have passed slightly before present recurrence. The sum comb filter, which is described by equation (4), can be rewritten for the case with weights:

TABLE 1
NON-ADJUSTED AND ADJUSTED COMB FILTER PARAMETERS

r	Non-adjusted filter first lobe attenuation, dB	Adjusted filter stop-band minimal attenuation, dB	a_1	a_2	a_3	a_4
3	10,99	11,90	0,64	0,20	-	-
4	13,01	13,27	0,72	0,44	0,12	-
5	13,86	13,98	0,68	0,60	0,36	0,12

$$y_k = \frac{x_n + \sum_{n=2}^{r} a_{n-1} \cdot x_{k-(n-1)D}}{1 + \sum_{n=2}^{r} a_{n-1}} \quad , \qquad (6)$$

where a is the weight for recurrence. The corresponding frequency response, similarly to equation (5), is given:

$$H(j\omega) = \frac{1 + \sum_{n=2}^{r} a_{n-1} \cdot e^{-(n-1) j\omega D}}{1 + \sum_{n=2}^{r} a_{n-1}} \quad . \qquad (7)$$

Each weight value, a, describes the amount of information, which is taken from previous recurrence, for the filter output calculation. In equation (4) all the weights are equaled with one.

Weight values change causes the change in filter frequency response shape. The filter weights can be selected according to the following determined parameters on frequency response plot: filter main lobe diameter on -3dB level, ripple amplitude in stop band, first side lobe attenuation and stop-band maximum magnitude value.

For comb filter, which extracts multiples of the fundamental frequency harmonical components, all previously mentioned parameters should be minimized. Unfortunately all parameters are related to each other. By minimizing one parameter, all the other parameter values are increased.

This article aim is to adjust the adaptive comb filter to the form, where it uses as few information from previous recurrences as possible. On the same time the adjusted filter frequency response should be as close as possible to the filter frequency response, which uses weight values a=1.

According to Table 1, the filters are compared with each other by the first side lobe attenuation. The adjusted filter whole stop-band magnitude should be at least as low as first side lobe magnitude of non-adjusted filter. By selecting the weight values according to previous conditions, the filter main lobe width is increased.

In this article, the filter weights a_n ($0<n<r$, where n and r are integer number) calculations are made numerically and

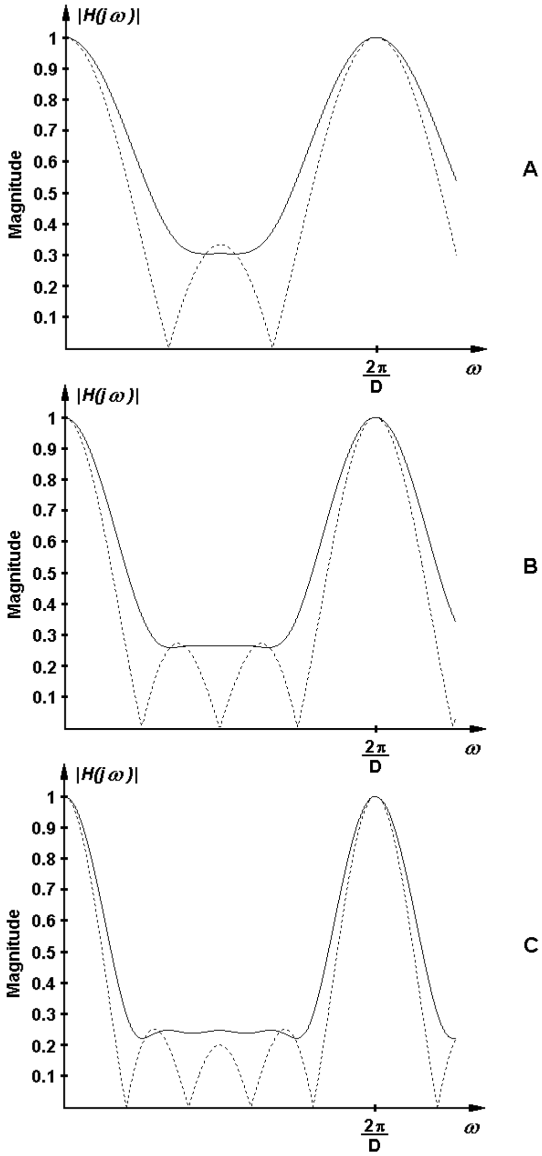

Fig. 2. Adjusted (solid line) and non-adjusted sum comb filter (dashed line) frequency responses in case D=8. A) Frequency response in case r=3. B) Frequency response in case r=4. C) Frequency response in case r=5.

changed between 0 and 1 with certain step s. The frequency response is calculated for all weight value combinations by using equation (7). Adjusted filter magnitude, in the place of non-adjusted filter first side lobe maximum, and stop-band maximum are calculated and separated into square matrices **L** and **M** respectively. **M** and **L** dimension equals with r-1, where r>2. In case the r=3, the **M** and **L** are $1/s \times 1/s$ matrices.

At first, all the values in **M** matrix, which are exceeding the allowed maximum, are eliminated. For the next step, the matrix is scanned through. The scanning here is explained for the case r=4, which means that **M** is 3D matrix. Matrix **M** elements are $m_{i,j,k}$. It must be mentioned, that $i \cdot s = a_1$, $j \cdot s = a_2$ and $k \cdot s = a_3$. The scanning starts, when i=0, j=0, k=0. The k value is changed until the end of matrix row. For the next step i=0 and j=1 and again the k value is changed until the end of matrix row. The previously explained scanning sequence is processed until the first existing matrix value is found, which was not eliminated before, and it corresponds to conditions i>j>k. Weights for adjusted filter are found and the scanning is finished, in case the **L** matrix value, in place of i, j and k, is less than allowed maximum. Otherwise the scanning is continued until the weights are found.

III. Results

The numerical calculations are made in MATLAB. The weight values are changed from 0 to 1 with step s=0.04. The matrixes **M** and **L** are calculated for filters with different r. All filter weights are found from matrices by using previously explained sequence. The results are given on Table 1. For comparison the adjusted filter minimal attenuation in stop-band is given respectively to non-adjusted filter first side lobe attenuation. In addition, the calculated weight values are given for adjusted filters. On Fig. 2 is given adjusted and non-adjusted filter frequency response plot.

As one condition, the adjusted filter stop-band attenuation had to be below non-adjusted filter first side lobe attenuation. On Table 1 and on Fig. 2 it is visible, that this condition has been fulfilled. For the filters on Fig. 2a and 2b the stop-band magnitude is almost straight, but on Fig. 2c the ripple in stop-band is visible. It is because the weights are determined with large step value. For better weight determination the smaller step value s should be used. Also the condition $a_1 > a_n > a_{r-1}$ is fulfilled.

The weight values in Table 1 are also describing the amount of information from previous recurrences, which is used for filter output calculation. By multiplying the weight values with 100, the amount of used information can be interpreted in percentages.

To compare adjusted and non-adjusted filters, which use different number of recurrences r, experiments were conducted. The PPG signal was generated by using one period length template. In addition reference signal was generated, which consisted of unit impulses. Each impulse marks the beginning of PPG signal period.

Generated PPG signal heart rate frequency was varied from 1Hz to 2Hz during 24 seconds. The noise was generated by using MATLAB random number generator and added to PPG signal. Four PPG signals with different SNR were generated.

Generated signals were filtered with three adjusted and three non-adjusted adaptive comb filters. Adjusted filters were using weights, which are given on Table 1. After filtering, the signal SNR was measured and noise attenuation was calculated.

The results are given on Table 2. Comb filter adjustment aim was to decrease the filter effect on signal shape

TABLE 2
NON-ADJUSTED AND ADJUSTED COMB FILTER USE OF INFORMATION FOR OUTPUT CALCULATION AND NOISE ATTENUATION

r	3	4	5
Adjusted comb filter			
Amount of information	184%	228%	276%
Noise attenuation	18dB	24dB	32dB
Non-adjusted comb filter			
Amount of information	300%	400%	500%
Noise attenuation	24dB	32dB	39dB

averaging by using less information from previous recurrences. The amount of information that filters are using from previous recurrences are given on Table 2. 100% corresponds to the situation, when one recurrence samples are involved to filter output calculation with weight 1. It is visible, that amount of information is decreased about two times. On the same time less information is taken from past recurrences.

The drawback of adjusted filter is the decrease of noise attenuation. For every filter the average SNR was calculated. It is visible, that adjusted filters have about 7dB better noise attenuation than non-adjusted filters by using the same number of recurrences for filter output calculation. It should be pointed out, that noise attenuation is equal for adjusted filter with *r* and non-adjusted filter with *r*-1.

IV. CONCLUSION

In this article adaptive sum comb filter adjustment was performed numerically for PPG signal. The past recurrence influence on currently processed recurrence was minimized, while keeping the frequency response properties as similar as possible for non-adjusted filter. The adjustment method was explained and new weight values were determined through numerical calculation. For adjusted and non-adjusted comb filter the comparison experiments were carried out with generated PPG signal. As future work, the analytical method for comb filter adjustment should be developed.

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