Piotr FIGOŃ, Paweł IRZMAŃSKI, Adam JÓŚKO
Warsaw University of Technology

ECG signal quality improvement techniques

***Abstract**. The problem of the presentation of ECG signals is discussed in the paper. The signal itself, beside the pure ECG content, often encapsulates artefact components with a physiological and technical origin. The latter element is the subject of Authors' research. The main goal is to develop dedicated filtering technique to reject low frequency artefact components including signal wandering as well as high frequency components with power line and radio interferences and noise.*

***Streszczenie.** W artykule przedstawione zostały zagadnienia związane z prezentacją sygnałów elektrokardiograficznych. W zapisie mogą wystąpić artefakty zarówno fizjologiczne jak i techniczne. Ta druga grupa zakłóceń jest przedmiotem badań Autorów. Głównym ich celem jest opracowanie techniki filtracyjnej, która pozwoli na usunięcie składowych zarówno wolnozmiennych (trend sygnału), jak i szybkozmiennych (zakłócenia interferencyjne sieciowe i stochastyczne). (**Techniki poprawy jakości sygnałów EKG**)*

Keywords: ECG signal automated analysis, Signal processing and denoising, Adaptive filtering
Słowa kluczowe: Automatyczna analiza sygnału EKG, Przetwarzanie i odszumianie sygnałów, Filtracja adaptacyjna

Introduction

Recently there have been observed permanent interest in the field of the automatic processing and analysis of biomedical signals[1,2,3]. The electrocardiogram is of course within the scope of the mentioned class of signals. The ECG signal reflects the electrical activity of the heart muscle observed from the body surface. ECG examination is a very meaningful and important element of cardiac diagnosis. It is possible to verify and evaluate normal cardiac function as well as distinguish a huge spectrum of symptoms and cardiac muscular dysfunctions, including serious pathologic cases head up with acute myocardial infarction.

Recorded ECG signal suffers from presence of complex sources of disturbances which deteriorate quality of desired pure ECG content and make it harder to observe and analyse and diagnose. That is the reason of implementing procedures responsible for elimination all noise sources without dispensable signal quality deterioration.

Disturbances and noise sources

ECG examination as all other measurement based process is distorted by noise. The origin of the noise is quite complex. The essential source of noise is the object - patient himself. This sort of disturbances is commonly called physiological artefacts. Breathing, muscle trembling, exudation and patient movement - these patient's activities introduce wide spectrum distortions which superimpose pure ECG signal. The frequency band of these artefacts is rather low as it extends form fractions of Hertz to few tens of Hertz (35 Hz for muscle trembling). Beside the group of distortions described above there is another specific variant of factors decreasing the value of SNR parameter. In turn, sources of these disturbances are all electric devices and in particular cardiosurgical equipment which causes electromagnetic interferences. The power system (mains) must be also taken into the account. Presence of these devices reveals compound and specific character of distortions. This time one discusses the technical artefacts. Their frequency parameters are quite different as the spectrum consists of different additional components. For the power system the frequency spectrum consists of 50 (60) Hz fundamental component and its harmonics. Electric devices interferences cover wide frequency spectrum including both lower and higher subbands (limited only by the sampling frequency and not excluding aliasing phenomena).

From the digital signal processing analysis point of view disturbances superimposing the clean physiological ECG signal cover wide signal classes of different frequency spectra (i.e. lowpass - breathing, highpass - electromagnetic interferences) and variable amplitude. For example electrical devices affect commonly results in interference noise amplitude much higher than the utile signal on the other hand power system interferences are rather lower than recorded ECG signal.

Each filtration technique, nevertheless it is a linear FIR/IIR, median or neural network one, it is aimed at removing distortion component without disturbing usable signal. But overly intensive distortion removal (filtering) may lead to the deterioration of components quality of utile signal (being analysed). On the other hand, the need to preserve as much as possible measurement information of useable signal may be the reason of no satisfactory disturbances filtering. Compromise must be fulfilled, then. As far as the electrocardiography signal is concerned, optimal filtering is a very difficult task to realise as the objectionable noise component nature is quite complex.

Consequently and continuously, adaptive filters technique is the one that has been gaining popularity recently. The class of DSP algorithms is based on parameters adapting variable conditions of function. Least Mean Square filters with or without memory and other variants, are those what are intensively used [4]. From the realisation point of view they are SOI (rather than NOI) filters with time variable frequency properties. Adaptive filters are especially useful for all cases where frequency spectra of noise disturbances superimpose useful signal. However, such filters require additional dedicated input, recording disturbance component only. As there are many leads being recorded during ECG examination, distinguishing disturbing constituent for a certain signal is not easy. On the other hand there is an additional right leg drive " noise" electrode. It might be used for noise properties estimation on the "software side", but the signal acquired this way it is not commonly available.

The adaptive properties of the algorithm presented in the paper is realised by means of local ECG signal parameters analysis, aimed at noise component estimation. Information on noise level is generated based on statistical research carried out with a set of reference database CSE. Results allowed for definition different stages of particular noise presence (50/60 Hz interference, 25/35 Hz muscle trembling) in the utile signal and development the dedicated set of SOI filters of time invariant coefficients. Due to the absence of "noise" input signal, there was a compromise agreed and simplified approach to adaptive filters was developed.

ECG signal SNR calculation algorithm.

Analysis performed for SOI filter stage selection starts at the point where SNR is calculated. This is done for each lead recorded independently.

Authors proposed and used two different methods for noise estimation. The first one is performed completely in time domain while the second one in frequency domain. The time domain approach is much simpler as it does not require DFT/FFT computations. Moreover complexity constraint was one of the essential project assumptions so that is why Authors decided to apply time domain approach.

Calculation procedure for determining SNR value is presented in the figure 1. An initial 2 seconds long ECG signal is divided into 10 equal 200 ms long segments. Two assumptions are agreed. First, at least one QRS complex is present during 2 s long ECG signal and the second, most of the 200 ms long segments do not contain any QRS complexes. Finally, obtained local amplitudes observed within each segment are compared to the “global” amplitude of full 2 s long signal. There are calculated global (squares) and local (circles) extreme presented in the figure 1. The lowest obtained value, pointed in green (fig. 1) is used for noise estimation operations.

SNR calculated in the way is not autonomous and independently offers no important information. It must be related to the greater number of specially selected, reference ECG signal set. As the reference database Authors used CSE Measurement set consisting of 10 s long, 12-leads, 125 signals. Additionally records included orthogonal Frank leads signals but they were not used. There was a research carried out to determine statistic, reference intervals of SNR changes for each lead individually. Based on this there was the decision algorithm proposed to select specific filter stage to be used for a particular signal.

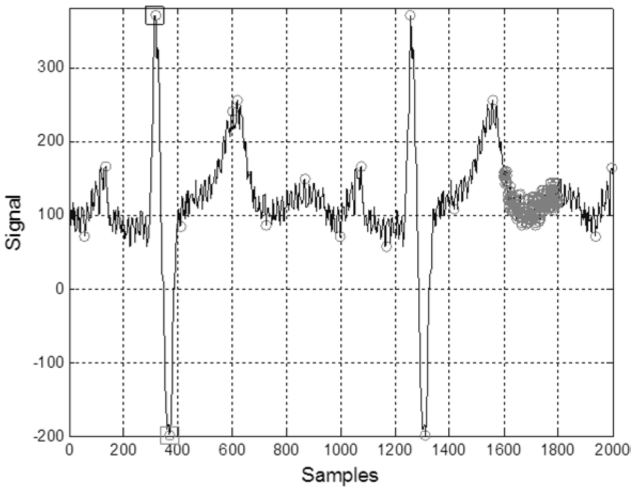

Fig. 1. ECG signal, dot - local minima and maxima on the basis of which the SNR is calculated, square - the global minimum and maximum

The decision algorithm

The principle of operation is based on the classification of the SNR factor to one of four numerical stages. Therefore on the function's input there is signal to noise parameter placed and on the output one have a value from 1 to 4.

As it was mentioned before the analysis of CSE database for particular leads was performed. The result of this operation is a collection of numerical values associated with each lead present in ECG record. Authors prepared twelve histograms calculated on 125 database files for each electrode). The figure 2 presents such data recorded in the V5 lead (the meaning of the red lines will be described later in this paper).

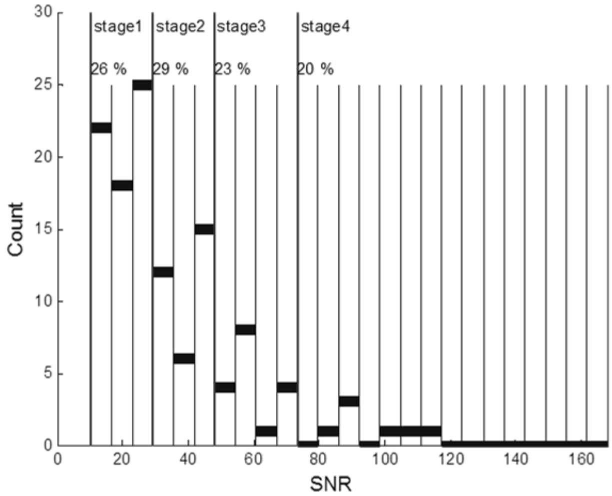

Fig. 2. Histogram presents SNR's number as a function of its value. The higher SNR the more noisy signal

The histogram is characterised by a negative asymmetry factor (see figure 2). It means that the SNR factors of the value 20%-30% of the maximum (associated with the most noisy signal) were the most frequent.

To obtain threshold values which indirectly are used to enable sequentially of digital filters, an area of the histogram was partitioned to four possibly equal parts. The red vertical lines are particular stages of filtering process (figure 2).

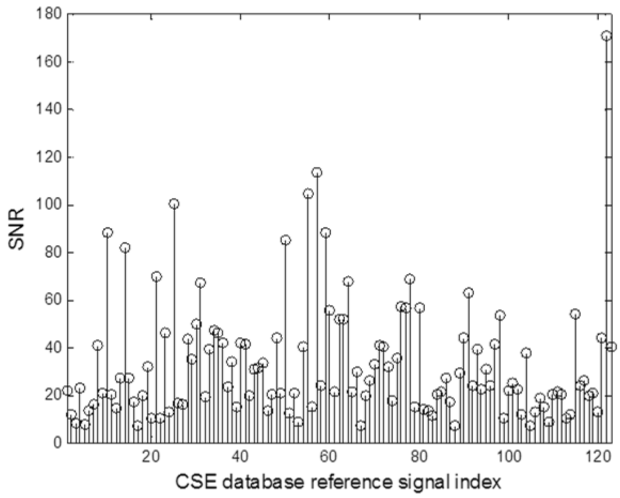

Fig. 3. A sample collection of SNR factors as a result of processing V5 lead

Digital filters

The core of the algorithm contains a set of four non adaptive FIR filters with constant coefficients. Those filters are non-recursive (no feedback from past outputs) and the major advantages of them are that the developing process is relatively simple, a linear phase is guaranteed and finally FIR filtering is always stable. On the other hand, the drawback of this DPS structure is a higher computational complexity in comparison to IIR filters (assuming the same effectiveness of filtration). As mentioned the embedded hardware has a limited CPU performance however Authors have decided to use the FIR filtering with maximum 20th-order and implementation of them in pure C++.

According to ECG record noise intensity four stages of filtering were used (presented below in detail). Going into details, each step is equipped with two FIR filters: the first one is used in online mode, the second one is for offline.

<u>1st Stage</u> - very high level of the disturbances, small value of the SNR. Enabling the 25Hz low-pass digital filter with the high attenuation ratio.
<u>2nd Stage</u> - applicable for high level of the disturbances, small value of the SNR. Enabling the 25Hz low-pass filter (as in the 1st stage) with a moderate attenuation ratio.
<u>3rd Stage</u> - used at average level of disturbances, average SNR value. Enabling the 35Hz low-pass filter with moderately high attenuation ratio.
<u>4th Stage</u> - low disturbance level, high SNR value. No filtration selected - the input signal in not modified.

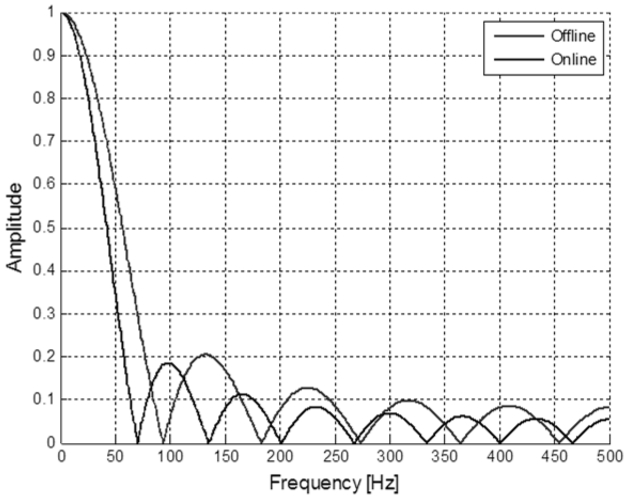

Fig. 4. Amplitude-frequency response of the filters used in the third stage (fs=1000Hz)

Described filters were developed using rectangular window technique. An example of described algorithm is presented in the following paragraph.

Algorithm application

There are two signals presented in different colours, in figure 5. The ECG record marked with black colour was randomly selected from the 1250 CSE database leads (MA1 folder containing artificial recordings) and used as the input signal. The processing was performed with use of all mentioned modules, the SNR calculation, the classification module and finally the filtering module with the strongest attenuation. The signal-to-noise ratio in that case was equal to 7.8. The result of the processing is marked with red colour. As one can see there is a significant improvement in the quality of the analyzed ECG record accompanied by a slight phase shift.

Summary

The presented paper describes simplified the auto-adaptation filter algorithm used to improve the quality of the ECG records and real data acquired from a patient. The structure was consisted of two crucial modules, the first one is designed to analyze input signal for the presence of disturbances, the second one is an four-level strength stage filtration method (FIR based in fact). Limited performance of the target platform prevents the implementation of algorithm based on frequency domain (DFT/FFT algorithm mentioned in the previous part of the document) but there is a possibility to use it in further hardware solutions.

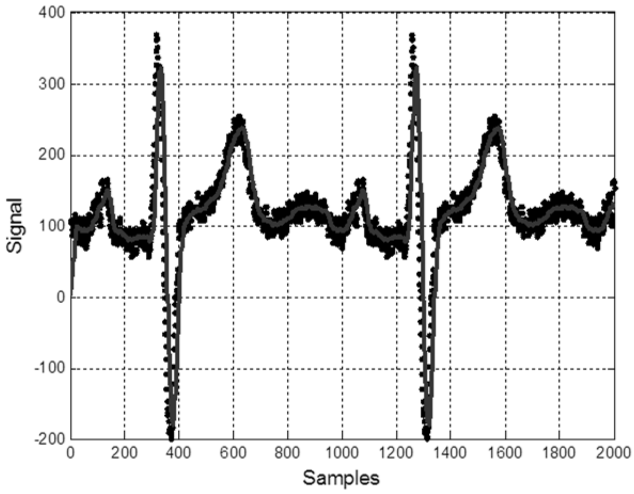

Fig. 5. An example use of the algorithm. The signal before filtration (blue) and the signal after filtration (red)

It was assumed, that the analyzed part of the ECG signal is representative for the rest of a record. It is correct for data length within range of a few tens seconds, however Authors cannot state that this is also true for longer ECG data records like Holter records (at the present moment).

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Authors:
Piotr FIGOŃ M.Sc., piotr.figon@ee.pw.edu.pl
Paweł IRZMAŃSKI M.Sc.,pawel.irzmanski@ee.pw.edu.pl
Adam JÓŚKO PhD.,adam.josko@ee.pw.edu.pl
Warsaw University of technology, Institute of Theory of Electrical Engineering, Measurement and Information Systems, 75 Koszykowa Str., 00-662 Warsaw.