ECG SAMPLING RATE CONTROLLED BY SIGNAL CONTENTS

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Abstract: This paper presents the ECG-dedicated algorithm performing the signal sampling at the variable rate. The local value of sampling interval is controlled by the bandwidth function representing the medical data density. The achieved data compression ratio equals 3 and the distortion level is very low. Practical considerations on adaptive anti-alias filtering and hardware implementation complete the paper.

Keywords: electrocardiography, data compression, non-uniform sampling

Introduction

Electrocardiogram is the most frequently performed electrophysiological test, indeed its accessibility is often limited by the cost of data transfer or storage. Despite the disscussion about the reliability of the restored signal, the compression of electrocardiogram data is of great practical significance and is widely used in clinical practice. An alternative solution, developed currently in our laboratory, makes use of variable density of diagnostic information in the cardiac signal that is obvious to the expert however difficult to express in a quantitative way.

The algorithm described in this paper continuously adapts the signal's sampling rate to the recording contents. Since full data bandwidth is necessary for short sections only (i.e. QRS complex), the use of variable sampling rate opens new possibilities of efficient data coding.

The algorithm uses the P, QRS and T wave borders detected automatically as reference points for the local signal contents. The general waveform of local importance function is next individually adapted to them. The pursuit of such function was the goal of our recent work [1], [2]. In the present implementation we used the function estimating global diagnosis quality loss as a consequence of local data cancelling in time-frequency domain [3]. It gives a quantitative measure of local signal importance in diagnostic aspect. This function controls the adaptive sampling rate algorithm in order to maintain best signal quality in its most important parts and economize samples of low importance in the neighboring zones.

Sampling the signal at variable rate involves two independent processes controlled by the bandwidth function: adaptation of anti-alias filters cut-off frequency and calculation of local sampling intervals. Both of them return quantization-free values in the continuous range from the minimum to maximum.

Today's offer of electronic devices makes the implementation of adaptive sampling rate algorithm feasible directly in the hardware. Digitally controlled anti-alias filters and converters triggered by countdown timers at any rate are widely available. The only constraint is the lack of methods for ECG segmentation accepting as input the variable sampling rate signal. For this reason our algorithm was first implemented in the software. The segmentation is performed by typical subroutine originally designed for an ECG recorder and next the constant sampling rate signal is transformed to its variable sampling rate equivalent. The original constant sampling rate is therefore the maximum rate for adaptive sampling.

Materials and Methods

The segmentation of an ECG signal may be performed by any subroutine, complying with the diagnostic standards [4]. For testing the variable sampling rate algorithm we used CSE Multilead database (2.44 μV, 500 Hz) that already provides reference segmentation points [5].

To estimate the required local bandwidth of the ECG signal we applied the temporal relationship of medical data vulnerability obtained by controlled data cancelling in time-frequency domain. The shape of this standard importance function is piecewise fitted into the current segmentation points of each heartbeat with use of cubic spline interpolation technique (fig. 1).

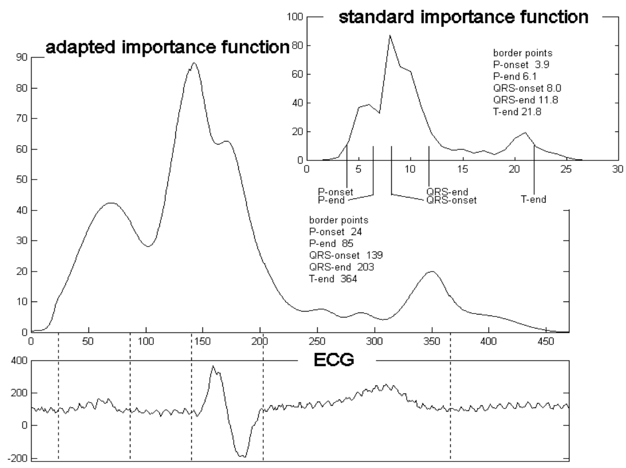

Figure 1. Standard importance function and its version adapted to the ECG signal (CSE, file Mo001)

In the software implementation of variable sampling rate algorithm the role of the digital anti-alias filter is to suppress all the components falling above the local bandwidth of the ECG signal and below the Nyquist frequency of the signal sampled at constant rate. For this purpose we adapted the sliding window average low-pass filter. The window's centre is moved to the consecutive samples of the original signal, but the window spans from 1 to 8 samples depending on the value of bandwidth function. The window length is not limited to integer number of samples, because border samples are partially included into the window with use of weighting coefficients.

Transformation of the constant sampling rate signal to its variable sampling rate equivalent begins with the computation of time points corresponding to irregular sample's positions. These positions depend on bandwidth function values (fig. 2). Next, the continuous ECG signal is simulated from regularly spaced samples. Finally, for each irregularly spaced sample the signal value is determined and memorised in the output data stream with use of cubic spline interpolation that here again was found the most appropriate technique.

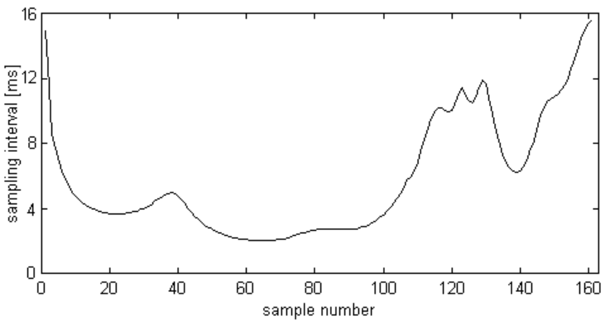

Figure 2. Sampling interval controlled by the values of bandwidth function (CSE, file Mo001)

Results

The algorithm of variable sampling rate implemented in the software was subject to the numerical test aiming to assess its performance. The average compression ratio (CR) is displayed in the table 1 together with the global and local differences estimate (PRD) between the original and the reconstructed ECG signals. Figure 3 displays the regular sampling rate signal and the variable sampling rate signal on a common time axis.

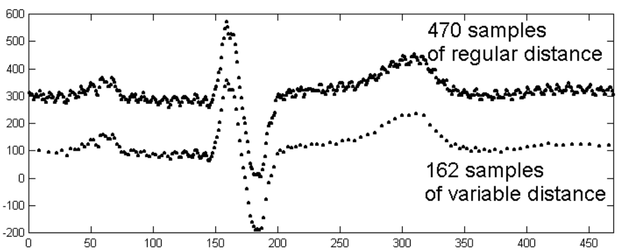

Figure 3. Comparing the heart beat represented in the regular and in the variable sampling rate signals

Table 1. The average compression ratio (CR) and differences (PRD) values.

CR		3.01
PRD [% (μV)]	global	3.11 (46.6)
	within P-wave borders	0.16 (2.4)
	within QRS-complex borders	0.22 (3.3)
	within T-wave borders	0.37 (5.6)
	out of waves	1.11 (16.6)

Discussion

The algorithm of variable sampling rate has been implemented in the software. The subroutine converts the constant sampling rate signal to its best variable sampling rate approximation and vice versa. The direct irregular sampling (hardware implementation) is also feasible, although requires the bandwidth function definition to be revised.

The compression efficiency is similar to the result of best lossless algorithm applied to the ECG [6]. Our method is in fact lossless from medical viewpoint, although it is not bit-accurate. The mean reconstruction error is negligible for the important parts of recording.

Main features of the variable sampling rate algorithm depend on the bandwidth function. The function estimating global diagnosis quality loss as a consequence of local data cancelling should be considered as proposal only and may be an initial point for further investigation or adaptation to the needs of particular users. The user-defined sampling profile is the principal advantage of this method.

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