Automatic Feature Extraction of ECG Signal Using Fast Fourier Transform

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Abstract

Electrocardiogram (ECG) is useful clinical information containing the condition of heart. The features of variations in ECG signal with time-varying morphological characteristics needs to be extracted by signal processing method because these are not easily visible in the conventional graphical presents of ECG signal. Large variations of simulated normal and noise corrupted ECG signal have been extracted using Fast Fourier Transform (FFT) method. The FFT method found to be successful in finding the abnormalities in ECG signal.

Keywords: ECG, FFT, DFT, z-plane, pole-zero location.

1 INTRODUCTION

Electrocardiogram (ECG) is a graphical record of the electrical activity that is generated by depolarization and repolarization of the atria and ventricles. It is well suited for analysis by joint time-frequency and time-scale distributions. ECG signal has a very time-varying morphology characteristic, identified as the P-QRS-T complex. The signal frequencies are distributed (1) low frequency - P and T waves, (2) mid to high frequency – QRS complex [1, 2]. When the heart muscle becomes ischemic or infarcted, characteristic changes are seen in the form of elevation or depression of the ST-segment. Ischemia also causes changes in conduction velocity and action potential duration, which results in fragmentation in the depolarization front and appearance of low-amplitude notches and slurs in the body surface ECG signals [3]. The statistical properties of ECG wave are generally changed over time tending to be quasi-stationary. A Holter monitor is an ECG recording done over a period of 24 or more hours. An automatic algorithm and software is needed to analyze this huge amount of 24 hours Holter ECG signals. A major problem is the proper detection of the ECG signal and extraction important features from it.

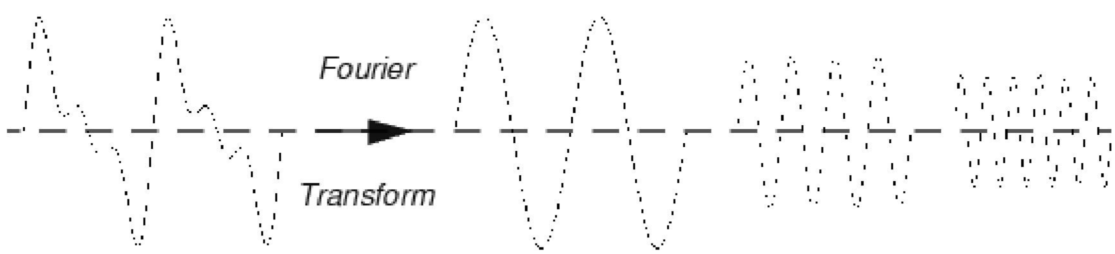
FFT methods have been used in a large number of biomedical applications. There is some works on precise detection of ECG using FFT [4-11]. Karel *et al*. proposed the performance criteria to measure the quality of a wavelet, based on the principle of maximization of variance [4]. Mahmoodabadi *et al*. developed and evaluated an electrocardiogram (ECG) feature extraction system based on the multi-resolution wavelet transform [5]. David *et al*. presented a method to reduce the baseline wandering of an electrocardiogram signal [6]. Shantha *et al*. discussed the design of good wavelet for cardiac signal from the perspective of orthogonal filter banks [9]. Nikolaev and Gotchev proposed a two-stage algorithm for electrocardiographic (EGG) signal denoising with Wiener filtering in the translation-invariant wavelet domain [10]. Most of the works focused on the large size abnormalities with respect to extreme noisy channel using conventional FFT and wavelet method. Most of the clinically useful information in the ECG is found in the intervals and amplitudes defined by its features (characteristic wave peaks, frequency components, and time duration). In this paper, the features of the ECG signal have been extracted using FFT methods. FFT method of signal processing is found to be superior to the conventional signal in finding the abnormalities in ECG signals.

2 METHODS

Mathematically, the process of Fourier analysis is represented by the Fourier transform:

$$F(\omega) = \int_{-\infty}^{\infty} f(t) e^{-j\omega t} dt$$

which is the sum over all time of the signal f(t) multiplied by a complex exponential. (Recall that a complex exponential can be broken down into real and imaginary sinusoidal components.) The results of the transform are the Fourier coefficients F(ω), which when multiplied by a sinusoid of frequency ω yield the constituent sinusoidal components of the original signal. Graphically, the process looks like fig.1.



Original signal Constituent sinusoidal components

***Figure1** Constituent sinusoidal components of the original signal*

Fourier analysis is extremely useful for data analysis, as it breaks down a signal into constituent sinusoids of different frequencies. For sampled vector data, Fourier analysis is performed using the discrete Fourier transform (DFT). The fast Fourier transform (FFT) is an efficient algorithm for computing the DFT of a sequence; it is not a separate transform. It is particularly useful in areas such as signal and image processing, where its uses range from filtering, convolution, and frequency analysis to power spectrum estimation. Working with the Fourier transform on a computer usually involves a form of the transform known as the discrete Fourier transform (DFT). The DFT is usually defined for a discrete function f (m , n) that is nonzero only over the finite region 0 <= m <= M-1 and 0 <= n <= N-1 . The two-dimensional M-by-N DFT and inverse M-by-N DFT relationships are given by

$$F(p,q) = \sum_{m=0}^{M-1}\sum_{n=0} f(m,n) e^{-j(2\pi/M)pm} e^{-j(2\pi/N)qn} \qquad \begin{matrix} p = 0,1,\ldots,M-1 \\ q = 0,1,\ldots,N-1 \end{matrix}$$

$$f(m,n) = \frac{1}{MN}\sum_{p=0}^{M-1}\sum_{q=0}^{N-1} F(p,q) e^{j(2\pi/M)pm} e^{j(2\pi/N)qn} \qquad \begin{array}{l} m = 0,1,\ldots, M-1 \\ n = 0,1,\ldots, N-1 \end{array}$$

The values F (p, q) are the DFT coefficients of f (m , n) . The zero-frequency coefficient, F (0 ,0) , is often called the "DC component." DC is an electrical engineering term that stands for direct current.

Here the ECG signal is used with the finite number of iteration. The sinusoidal periodicity is definite with the assist of Dirichlet condition. The Dirichlet conditions are as follows.

1. The ECG signal has a finite number of discontinuities in any period.
2. The ECG signal contains a finite number of maxima and minima during any period.
3. The signal is absolutely integrable in any period.

The FFT method uses normally DFT method to satisfy the procedure to plot the signal in frequency domain both in frequency response and phase response. The L-point DFT is sufficient to uniquely represent the sequence of the discrete signal in the frequency domain, it is apparent that it does not provide sufficient detail to yield a good picture of the spectral characteristics of the signal. If we wish to have better picture, we must interpolate the frequency response at more closely spaced frequencies. In fact, we can view this computation as expanding the size of the sequence L points to N points by appending N-L zeros to the discrete sequence, that is, zero padding. Then the N-point DFT provides finer interpolation than the L-point DFT.

The continuous time domain ECG signal has been distorted to discrete time domain signal using sampling theorem. The discrete values and the DFT point have been taken to be used in the DFT process. Here the discrete values n are taken in terms the DFT point of m which are belonging to the above two DFT equation. The parameters p and q hold the DFT coefficients that can help to draw the plot of frequency response and the exponential part which carries the phase that can help to draw the plot of the phase response of the ECG signal.

3 RESULTS AND DISCUSSION

The simulated standard ECG signals as well as the simulated noise corrupted ECG signal have been generated using MATLAB. From the human body, sudden pain of any parts may occur the continuous sinusoidal signal with frequency with approximately 1/5 Hz cause the abnormalities of the cardiac activities of heart. Signals have been generated with different parameters using the following steps.

Step 1: Generation of standard ECG pattern having amplitude of 3.5mV and pulse repetition rate of 75 per minute. This signal is shown in Fig.2(a).
Step2: Generation of a noisy signal having frequency of 1/5 Hz and amplitude of 1.5 mV which is approximate 40 percent of the standard ECG signal. This signal is shown in Fig.2(b)
Step 3: Features are extracted of the signal using FFT method.

The dissimilarities of normal and noise corrupted ECG signals are needed to record long time for proper diagnosis. Figure 2 shows the FFT output of normal ECG signal and noise corrupted ECG signal. The dissimilarities are identified by the parameters such as amplitude, frequency, phase and pole-zero plots. There are some differences between the parameters of simulated standard and noise corrupted signal which diagnosis the proper treatment for the patient.

Every digital filter can be precised by its poles and zeros (together with a gain factor). Poles and zeros provide functional insights into a filter's response, and can be used as the basis for digital

filter design. It additionally presents the Durbin step-down recursion for checking filter stability by finding the reflection coefficients, including matlab code.
We can write the general transfer function for the recursive LTI digital filter as

$$H(z) = g\frac{1 + \beta_1 z^{-1} + \cdots + \beta_M z^{-M}}{1 + a_1 z^{-1} + \cdots + a_N z^{-N}}$$

In the same way that $z^2 + 3z + 2$ can be factored into (z+1)(z+2), we can factor the numerator and denominator to obtain

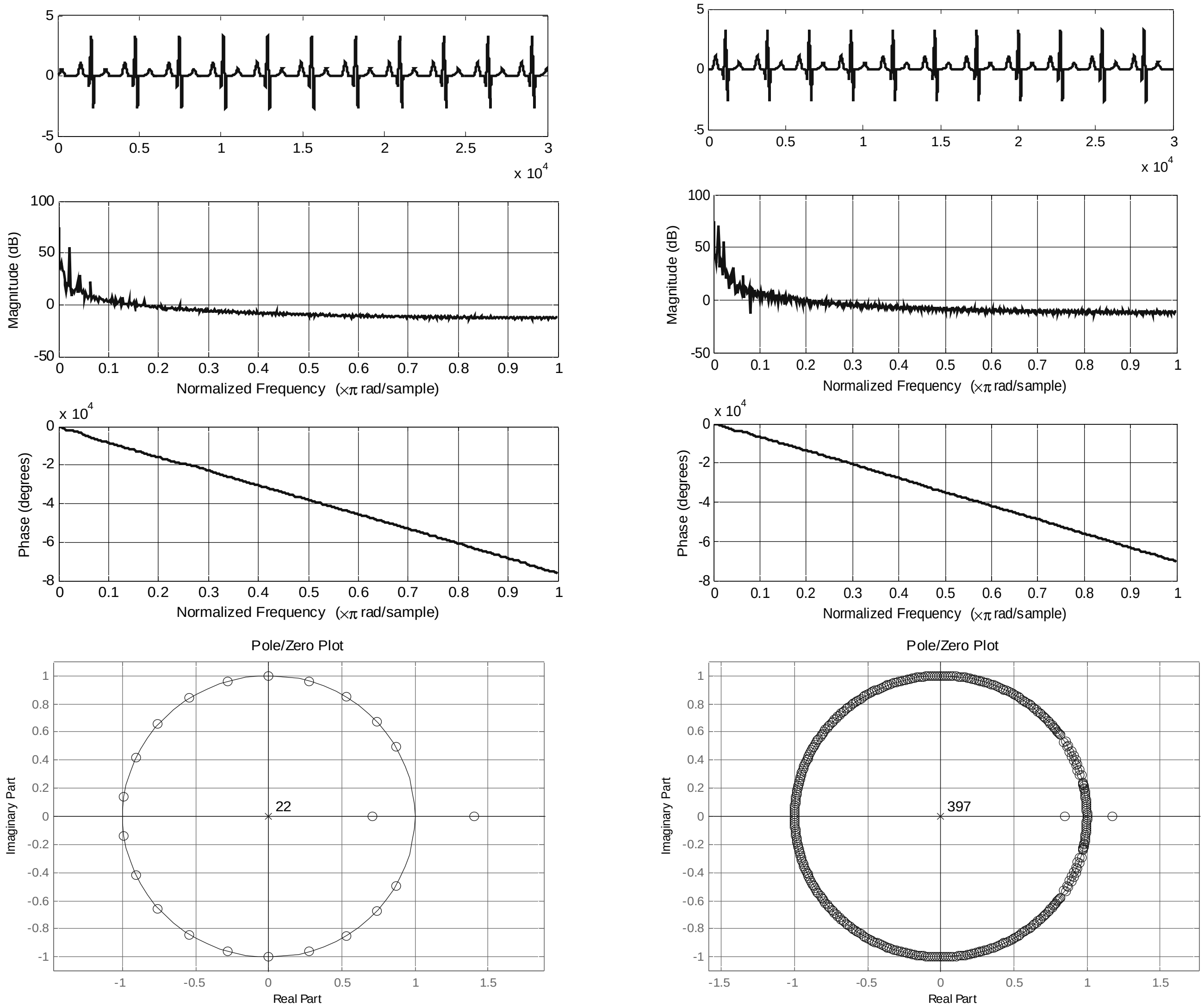
$$H(z) = g\frac{(1 - q_1 z^{-1})(1 - q_2 z^{-1})\cdots(1 - q_M z^{-1})}{(1 - p_1 z^{-1})(1 - p_2 z^{-1})\cdots(1 - p_N z^{-1})}.$$

Assume, for simplicity, that none of the factors cancel out. The (possibly complex) numbers {q1, . . , q_M} are the roots, or zeros, of the numerator polynomial. When z is set to any of these values, the transfer function evaluates to 0. For this reason, the numerator roots q_i are called the zeros of the filter. In other words, the zeros of the numerator of an irreducible transfer-function are called the zeros of the transfer-function. Similarly, when z approaches any root of the denominator polynomial, the magnitude of the transfer function approaches infinity. Consequently, the denominator roots { p_1, . . , p_N } are called the poles of the filter.

The term ``pole'' makes sense when one plots the magnitude of H(z) as a function of z. Since z is complex, it may be taken to lie in a plane (the z plane). The magnitude of H(z) is real and therefore can be represented by distance above the z plane. The plot appears as an infinitely thin surface spanning in all directions over the z plane. The zeros are the points where the surface dips down to touch the z plane. At high altitude, the poles look like thin, well, ``poles'' that go straight up forever, getting thinner the higher they go.

Notice that the M+1 feedforward coefficients from the general difference Equation, give rise to M zeros. Similarly, the N feedback coefficients give rise to N poles. Recall that we defined the filter order as the maximum of M and N. Therefore, the filter order equals the number of poles or zeros, whichever is greater.

Standard ECG signal is sinusoid in time domain with the amplitude of 3.5 mV periodically. If we add sinusoidal noise with the amplitude of 1.5 mV including the $1/5^{th}$ of the frequency of standard ECG, the noise corrupted ECG signal behalves varying the signal amplitude and the frequency response. From the pole-zero definition, we see the status of the signal. When standard ECG signal shaped the z-plane using pole-zero location, the system is stable (all poles are in the unit circle) and the finite number of zeros are exist. But when the noise corrupted signal is taken, the system is also stable, it has like infinite number of zeros which carries the information that extra signal energy is being resided and may hamper the original signal reconstruction.



a) **b)**

Figure 2 *Comparison between normal (a) and noise corrupted (b) ECG signals using FFT*

4 CONCLUSIONS

In this paper, simulated normal ECG and simulated noise corrupted ECG signal conditions have been analyzed using conventional FFT method. The abnormalities of cardiac parameter are detected using FFT, which were not visible in the conventional graphical presents of ECG signal. The frequency response and the pole-zero locations have been considered to evaluate the signal differences between simulated normal and noise corrupted ECG signal. And the changes have been found to detect the desired results. The FFT method found to be more precise the abnormalities in ECG signal.

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