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The Analysis of Heart Sounds and a Pocket Computer Application via Discrete Fourier Transform

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1. Introduction

The heart is one of the two crucial centers for human life. Thus, any disorder concerning the heart which may be able to occur is of prime importance to human health. The mortality rate stemming from heart diseases took the second lead after brain embolism in the world from 1985 to 2006 (Jiang & Choi, 2006). Being such significant to humans, the heart organ is consisted of two cycles. At the moment of the closure of mitral and tricuspid valves, ventricular contraction comes out, alias one cycle begins with systole and ends with diastole (Sharif et al., 2000). Auscultation with stethoscope is a preferential method that the doctors use in order to differentiate normal cardiac systems from the abnormal ones that come out (Sinha et al., 2007). The listened heart sounds are formed through the flow of blood entering and exiting the heart and with the movements of cardiac valves connected to this flow. The sounds comprised of this blood flow are listened by the physicians via stethoscope. The heart sounds are interpreted and it is determined whether the patient has any disorder about the heart. On the other hand, auscultation method has some limitations. Auscultation depends on the physician's interpretation of different heart sounds, hearing skill, experience and expertize (Kandaswamy et al., 2004). The required experience and expertize are achieved as a result of long examinations and diagnoses made by the physicians. Even though one physician has the necessary training so as to conduct auscultation and to diagnose cardiac disorders, he is still in need of clinical experience. Lacking experience and expertize are especially a difficulty for the newly graduates and medical interns. However, unsuitable conditions of the environment and incompatibility of the patient can also lead to deficiency in the diagnosis process. Because of these difficulties which may be experienced, auscultation method that is listening with a stethoscope has been insufficient in the exploration of heart abnormalities.

As the auscultation method fail to meet the needs of the physicians, they also make use of Electrocardiography (ECG) data along with stethoscope. ECG method, generated through the advancement of the technology, is a procedure helping physicians in the diagnosis of the

heart disorders. ECG is the wave form taking the record of the electrical activity of the heart via electrodes attached to the skin. A simple and cheaply method, ECG is constantly used by the physicians. ECG records and their analysis that are used to detect defects in the heart are relatively a good method. On the other hand, if the likely generating system of heart is utterly acting with small heart defects, it may look very difficult to diagnose these abnormalities with the analysis of ECG records as nearly no change in the ECG will be discovered. However, it was determined that this situation has almost led to changes among the sounds produced by the heart (Sinha et al., 2007). As it is not possible to detect some heart abnormalities with ECG, listening to heart sounds with auscultation method has earned more importance. Owing to the reasons explained above, there is a need for faster and more accurate diagnosis in the record and analysis of these heart sounds.

The very first phase of developing the system that will help the physicians interpret the heart sounds accurately is the signal processing methods. When the studies based on these methods are searched, it can be seen that Fourier analysis has been used in many of the studies. Thanks to the Fourier analysis, it is possible to examine sound signals in the frequency space. In studies especially done on the classification of heart sounds, the frequency analysis of these digitized heart sounds are processed and then passed to classification stage with various artificial intelligence methods. If we survey some of the studies in literature that used heart sounds and Fourier transform together, in one study Segaier et al. developed a digital algorithm in order to detect the first heart sound (S1) and the second heart sound (S2) and to characterize systolic murmurs. The study done on the pediatry patients, Short-Time Fourier Transform (STFT) was used so as to carry out the analysis of the heart sounds taken from the patients (El-Sagaier et al., 2005). In another study, Folland et al., in the moment of auscultation, applied Fast Fourier Transform (FFT) and Levinson-Durbin algorithm to heart sounds to analyse the abnormalities in the heart sounds; and applied the data to the Multilayer Perceptron (MLP) and Radial Basis Function (RBF) artificial neural networks to classify abnormal sounds. Eventually, it was determined that in the classification of the heart sounds, the sensitivity levels that the MLP and RBF neural networks have gained were 84% and 88% consecutively (Folland et al., 2002). In another study, Debbal ve Breksi-Reguig have done the time-frequency analysis of S1 and S2 heart sounds, and applied Wigner distribution and wavelet transform techniques to the heart sounds. They made a comparison between FFT and STFT and with these techniques they implemented (Debbal & Breksi-Reguing 2007). In one of the study by Abdel-Alim et al., they made a classification of heart sounds that belonged to different cardiac valve disorders with the use of a feed forward artificial neural network. In the course of analysis of these sounds, discrete wavelet transform, FFT, and linear prediction coding methods were employed. In the end, the classification success was found out to be %95.7 (Abdel-Alim et al., 2002).

In this study, the heart sounds achieved through a stethoscope were initially computed, and they were subjected to Discrete Fourier Transform (DFT) and next the graphics and the frequency spectrum in the time domain that belonged to the heart sounds were drawn on the pocket computer. Thus, frequency spectrum of normal and abnormal heart sounds was obtained via DFT, so it was aimed to prepare the physicians with some more data in the course of auscultation.

2. Structure of the heart and heart sounds

Heart is an empty muscle that pumps the blood to the blood vessels in the whole body (Sharif et al., 2000). The most significant and primary duty of the heart is to dispatch and pump the blood to the circulation system like a pump (Ahlström, 2006).

As it can be seen in figure 1, the heart is composed of two parts: the right heart and the left heart. The right heart pumps the blood into lungs. This cycle is called as pulmonary cycle. On the other side, the left heart is the part that provides all the organs and the whole body with the oxygen and nutrients (Sharif et al., 2000). In addition, there are four chambers in the heart that are known as the right and the left atriums and ventricles. These two atriums are the places where the blood entering the heart is stored. The ventricles, on the other hand, convey the blood to the whole body like a pump. As the heart contracts, the blood makes a pressure towards the valve and moves from the atrium to the ventricle (Barschdorff et al., 1990).
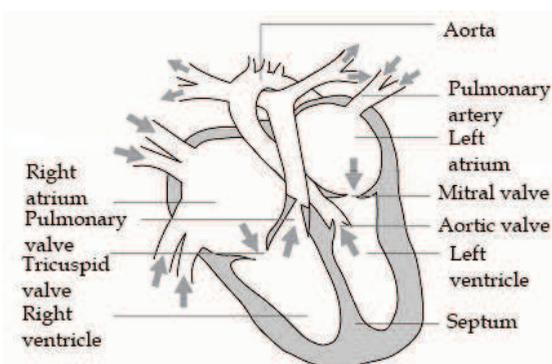

Fig. 1. Structure of the Heart

The relation among the volume, pressure and flow of the blood in the heart determines the opening and the closing of the valves. Normal heart sounds occur in the course they close. Besides, the sounds occurring in the heart and in the vessels with the flow are constituent of the heart sounds, too. But, as a matter of fact, how they occur is still a matter of discussion (Ahlström, 2006).

The abnormalities in the heart structure mostly reflect to the sounds that the heart produce (Leung et al., 2000). The formation of heart sounds and murmurs are generally developed through the actions of myocardial walls, the opening and closing of cardiac valves, and the blood flow into and out of the ventricles (Kemaloğlu & Kara, 2002).

While the sinoatrial node contracts itself and the contraction spreads to the atria, the pressure of left atrium surpasses the pressure of left ventricle. This contraction is then extended to the ventricles. This moment is the time when ventricle starts to depolarize. The contraction spreading into the ventricle contracts the muscles of the ventricle and leads to a contraction in the ventricle. The pressure in the left ventricle starts to rise and as soon as it reaches to the pressure of the left valve, the atrium and the valve ventricles close. At that moment, S1 comes out. Mitral valve normally closes earlier than the triscuspid valve. For that reason, S1 has two elements namely mitral and triscuspid. The frequency band is 20–45 Hz and the period is 50–100 ms.

The ventricle pressure goes on rising, and as the pressure goes over the pressure of aorta, firstly aorta valve and next pulmonary valve open. Soon after starts the period of sending the blood out of ventricles. As the ventricle muscles relax, the ventricle pressure starts to decrease. At the moment that the internal pressure in the ventricle goes below the aorta pressure, the aorta valve closes. The pulmonary valve closes respectively. The closure of these two valves forms the S2 sound. The frequency band is 50–70 Hz and the period is 25–50 ms.

When the internal pressure of the ventricle drops below the pressure of atrium, mitral and triscuspid valves open and the ventricles are filled with blood. As the ventricles are filled with blood, the vibrations of the ventricle muscles form the third heart sound (S3). It could be heard in the young while it is an indicator of disorder of myocard function in the overaged. This sound arises almost after 150 ms after the closure sound of aorta.

In the last parts of blood fulfillment of the ventricles, the flow of blood that was again quickened with depolarization of the atrium revibrates the ventricle walls and in some pathologic cases, this causes the fourth heart sound (S4). S4 is normally not heard in the adults but could be taken in children.

Four sounds seen in figure 2 are known as the simple heart sounds. Apart from these sounds, some sounds as murmur may occur in some heart disorders. These sounds are in the frequency band of 100–600 Hz and are long-time compared to the simple heart sounds (Kemaloğlu & Kara 2002).
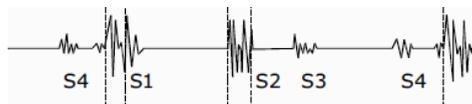

Fig. 2. The First, Second, Third and Fourth Heart Sounds

The sounds that are named as murmur and caused by the blood passing through the cardiovascular system loudly are the significant examples of abnormal sounds. The timing of the murmur and the level of height have remarkable significance concerning the situation of the heart. For instance; in the course of diastole a murmur marks erroneous functionality of the heart valve. On the other hand, in the course of systole, the murmurs may be related with the healthy and pathological heart depending on the acoustic character of the murmur (Ölmez & Dokur, 2003).

The murmurs are formed through irregular blood flow and as a result of narrowing and leaking valves or the existence of the abnormal passages in the heart. Eventually, the blood flow that comes out leads to steady and irregular vibrations that are transmitted from the cardiac and tissues belonging to the chest to surface of the chest (Ahlström, 2006).

While defining the heart sounds, we need to draw attention to the frequency, density and the quality of the sound. For that reason, it is a must to listen to the heart sounds carefully that are used as reference S1 and S2 and to determine the place accurately in the course of listening. These sounds experience the systole and diastole phases as the heart operates, and the sound differentiations in these places provide related information in connection with the disorders in the heart (Say, 2002).

3. Fourier transform

When the signals in the real world are searched, it could be said that the signals that are encountered practically are the time domain signals and the size measured is the function of the time. For that reason, the signal needs to be transferred to a different domain through an application of mathematical transform and some information from the constituents that represent the signal in the domain can be obtained about the signal. For example, the frequency spectrum (frequency constituents) of the signal is gained with Fourier transform. The information hidden in the time domain is brought out in the frequency domain (Say, 2002).

The Fourier display of the signals plays an extremely substantial role in signal processing in both continuous time and discrete time. Thus, it supplies a method to match the signals to a domain and makes it possible to study on them. Fourier transform allows for a distinctive way to interpret the signal and systems (Hayes, 1999).

Fourier transform is a method of analysis that was developed by Jean B. Joseph Fourier, a French physicist and a mathematician, in 1987 when he was studying on the Fourier's research about heat and dispersion and it plays an important part in the signal processing (Swanson, 2000). In the study that Fourier conducted in 1807, he said that it could be achieved through selecting and gathering the sinus and cosine waves among the continuous and periodical signals (Smith 1997).

The Fourier transform that is used to determine the frequency constituent of the raw signal in the time domain can be defined with the two equities as follows:

$$X(\omega) = \int_{-\infty}^{\infty} x(t).e^{-j\omega t}.dt \tag{1}$$

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\omega).e^{j\omega t}.d\omega \quad \omega = 2\pi f \tag{2}$$

With Fourier transform, the signal is extricated into complex exponential functions that have various frequencies. In the equations t, specifies time; ω, specifies angular frequency and f specifies the frequency. x, specifies the signal in the time domain, X specifies the signal in the frequency domain. In the equation 1 given above, the Fourier transform of $x(t)$ and in the equation 2, inverse Fourier transform are displayed.

When the equation 1 is viewed, it is seen that $x(t)$ signal is multiplied with an exponential term in a certain f frequency and the integral of this multiplication is taken from minus infinity to plus infinity all over the time. It should be taken into account that the effect of the f frequency constituent to the integration will be the same no matter when it may come out among these times. The integration result will not change whether the f frequency constituent comes out in the course of $t1$ or $t2$. Fourier transform only indicates whether a certain frequency constituent exists or not. Just a spectral constituent of a signal can be gained through Fourier transform (Say, 2002).

Today, given that, the signal processing is held by computer algorithm and the computers may work in limited length and with discrete signals, it is noticed that the Fourier transform to be used will be discrete time Fourier , in other words, DFT (Smith, 1997).

3.1 Discrete Fourier Transform

On the contrary to some arrays defined theoretically, the Fourier transform of real arrays may not be calculated. Thus, it is not convenient for the digital signals to use Fourier transform. That the frequency is displayed analogically and it requires infinite number samples are some of the basic reasons of this ineptitude.

Due to these difficulties, if the importance of Fourier transform in signal processing is taken into account, a more practical transform needs to be defined. The new transform defined as normal spaced N frequency point (Ωk) around a unit disc and N sample of $x(n)$ arrays is called as DFT (Kayran & Ekşioğlu, 2004).

DFT is in essence a kind of transform such as Fourier arrays transform and Fourier integral transform. The transform feature is very powerful for the time arrays, which also allows for inverse transform. As can be noticed from its name, it owns utterly similar characteristics with Fourier integral transform. It especially defines the spectrum of a time array (Cochran et al., 1967).

This transform that also allows inverse transform includes distinctive qualities. The principal quality is the equivalent of multiplication of two DFT in the time domain is the total convolution of arrays. In addition, many spectrum analysis methods are based on DFT (Kayran & Ekşioğlu, 2004).

DFT is defined through the equation 3 given below:

$$A_r = \sum_{k=0}^{N-1} X_k \exp(-2\pi jrk/N) \quad r = 0,...,N-1 \tag{3}$$

Here, A_r, symbolizes rth coefficient of the DFT and X_k symbolizes kth sample of a time arrays composed of N sample. X_k's may be complex numbers while A_r's are always complex numbers. The formation of the formula that is consistent with the notation given in the equation 3 is mostly shown with the formula given in the equation 4:

$$A_r = \sum_{k=0}^{N-1} (X_k) W^{rk} \quad r = 0,...,N-1 \tag{4}$$

$$W = \exp(-2\pi j/N) \tag{5}$$

Since X_k's are commonly the values of discrete time points of a function, r arrays are sometimes referred as the frequency of DFT. DFT, too, is termed as "discrete time, finite range Fourier transform".

If we take a look at the inverse transform of equation 4:

$$X_l = (1/N) \sum_{r=0}^{N-1} A_r W^{-rl} \quad l = 0,...,N-1 \tag{6}$$

This relation (equation 6) is termed as the inverse of DFT (Cochran et al., 1967).

In the figures between 3 and 7 below, the flow diagram of DFT algorithm that was taken as a basis in the programming of the pocket computer was given. The main module seen in figure 3 calls for four modules, which are used in the calculation process.

The next module, shown in figure 4, sets up the twiddle factor arrays, that is the computation and storage of the sample values of cosines and sinus over one cycle, with argument starting from zero with increment of $2\Pi/N$. This module is named as *twid_fac*. *i* variable, here, is used as loop counter. On the condition that *i* variable reaches N that is the data size, the loop is terminated. The *tfc* and *tfs* arrays whose sizes are equal to N are used to hold the sample values of sinus and cosines respectively.

The next module seen in figure 5 reads the real and imaginary parts of the complex input data, and in turn transfers them into N dimensioned *xr* and *xi* arrays. It is assumed that the real parts of the input data are stored before the imaginary parts in the input file. If the data is real, the data in the *xi* array is initialized to *0* and the data in the *xr* array is read. This module is called *in-put.*
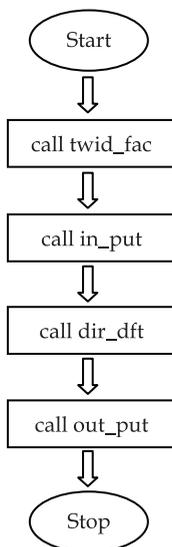
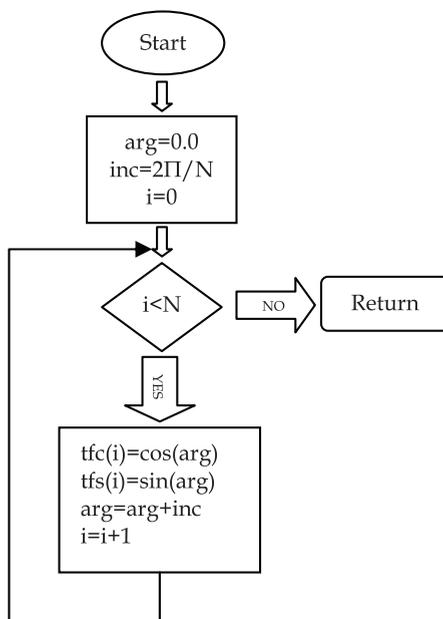Fig. 3. The Main Module for Direct Implementation of the DFT
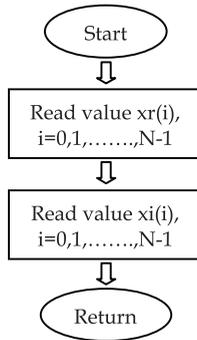
Fig. 4. The Twiddle Factor Module

Fig. 5. The Input Module

The next module that is *dir_dft* module is given in figure 6. *dir_dft* module is used in the calculation of DFT coefficients. The calculation has been achieved via two nested loops. While the outer loop controls the frequency index, the inner loop controls the access of the data values. In each outer loop's iteration only one coefficient is calculated. The real and imaginary parts of the coefficients are stored, respectively, in arrays *XR* and *XI*, each of size *N*. The access of correct twiddle factor values is carried out using the *mod* function. Inside the inner loop, each coefficient is computed according to the DFT definition.
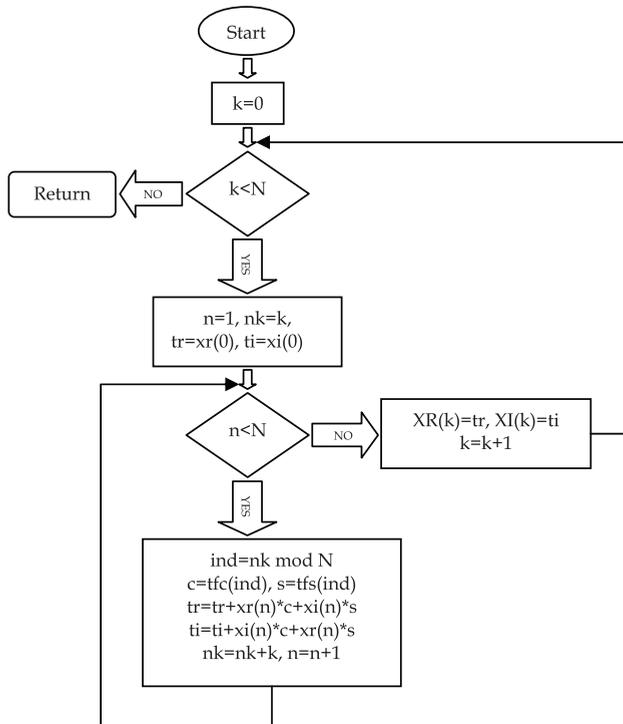

Fig. 6. The DFT Module

The last module, shown in Figure 7, prints the real and imaginary parts of the coefficients, respectively, from the arrays *XR* and *XI,* one coefficient in each iteration. This module is called *out-put (Sundararajan, 2001).*
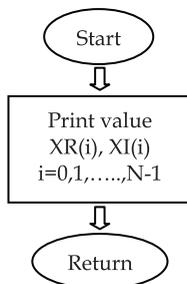
```
        ┌─────────┐
        │  Start  │
        └─────────┘
             ⇩
   ┌──────────────────┐
   │   Print value    │
   │   XR(i), XI(i)    │
   │  i=0,1,…..,N-1    │
   └──────────────────┘
             ⇩
        ┌─────────┐
        │ Return  │
        └─────────┘
```

Fig. 7. The Output Module

DFT calculations are used in many scientific and engineering applications (Winograd, 1976). For the effective and advantageous use of DFT, some of its basic properties need to be learnt. It is beneficial to talk about its features shortly. The first characteristic of DFT is the linearity feature. DFT is a linear transform. Another feature of its is the symmetry feature. DFT values that are made of real values and equal to a periodical array are complex and periodical. One more feature is the principle of similarity concerning the selectivity of time and frequency. The indefinity principle of DFT is connected with the terms in the time and frequency domain of DFT. It is equal to the well-known indefiniteness principle in physics. This term is not the result of physical properties, but it is the result of a chief mathematical formulation. The final feature is the conditions of link-equivalence between DFT and Fourier Transform. Since DFT is the approximate of the continuous Fourier transform, researchers are concerned with it. The validity of this approximation is certainly based on the related wave form (Kayran & Ekşioğlu, 2004).

4. DFT application on pocket computer

In this study, the frequency analysis of the heart sounds taken from the patient via electronic stethoscope was actualized, and on the pocket computer it was displayed both in the time and frequency domain. Just as the system was completely developed mobile, it was aimed that the physicians may use it in the course of clinical examination. The physician using the system in the course of clinical examination will be able to make a more accurate and fast diagnose through listening to the heart sounds and following the heart sounds both in the time and frequency domain on the device.

4.1 Digitization of the sound
In order to play and store the heart sounds on the computer media, we need to have some kind of file formats. Because of these formats, multimedia file can be listened and stored on the computer. The *wav* (Waveform Audio Format) format to be used in our study is a kind of sound file. It is a commonly used sound file format. Unlike the other sound formats, the sound in the *wav* file format cannot be stored by compressing; it can only be stored through digitizing. As this file format is not compressed, it holds a lot of space. On the other hand, the sound quality is quite good. In this study, the *wav* sound format has been selected as it saves the digitized sound with its original state before compressing.

One *wav* file is in general consisted of two parts. In the first part, there is general information concerning the data. The second part is the part where the original data starts.

In figure 8, the structure of a *wav* file format in general has been given (a. Güraksın, 2009).

Field Name	Field Size in Bytes
Chunk ID	4
Chunk Size	4
Format	4
Subchunk1 ID	4
Subchunk1 Size	4
Audio Format	2
Num Channels	2
Sample Rate	4
Byte Rate	4
Block Align	2
Bits Per Sample	2
Subchunk2 ID	4
Subchunk2 Size	4
Data	Subchunk2 Size

The format of concers here is "WAVE" which requires two sub-chunks: "fmt" and "data

Describes the format of the sound information in the data sub-chunk

Indicates the size of the sound information ad contains the raw sound data

Fig. 8. The Canonical *wav* File Format

The heart sounds taken from the patient are digitized through a piece of code written accordingly to the mentioned format. During the digitization process, the digital data that are in the hexadecimal basis are obtained by reading *wav* format and are then converted into decimal basis. Thanks to the obtained digital format of the sound, it is made possible to draw a graphic of the sound on the pocket computer and DFT can be applied to the sound. With the help of a software that operates on the pocket computer, the digital data obtained from the wav sound format are turned into sound graphic in another code, and again with the help of another programming code it is subjected to DFT.

4.2 Raw data obtainment

The sounds used in this study were taken by using Littmann 4100 model electronic stethoscope from Afyon Kocatepe University with the 07.AFMYO.01 numbered scientific research project (a. Güraksın, 2009). With model 4100 Littmann electronic stethoscope, it is possible to record 6 different heart sounds. Thanks to this, the sounds taken continuously from six patients are stored within the stethoscope itself. In addition, the saved sounds will be able to be transferred to pocket computer with the help of infra-red technology within the stethoscope. The sounds recorded with Littmann 4100 model electronic stethoscope are

stored in *e4k* format. This format was converted into *wav* format on the pocket computer via a program given by Littmann. After converting the heart sounds into *wav* format, we pass to the next phase of signal processing. By and large, there are some stethoscopes in the market that are able to convert the stored sounds directly into wav format. By using such a kind of stethoscope, there will be no need to use a second transform. In order for the sounds taken by stethoscope to be processed, HP iPAQ hx2000 pocket computer that allows for infrared transaction was used.

4.3 Drawing DFT graphics of the data of heart sounds on pocket computer

The aim of this study is to get the frequency domain graphic that was obtained by DFT and the time domain graphic of the heart sounds taken from the patients drawn on the pocket computer. The software developed for this purpose was prepared through the use of programming language of C# in Microsoft Visual Studio 2005 media.

As you can see in figure 9, firstly the sound was taken from the patient by using the Littmann 4100 model electronic stethoscope. Next, the sound to be analysed in a program written in Visual Studio 2005 media was digitized, and it was subjected to DFT method. In the final stage, the graphics of the processed heart sounds both in the time domain and frequency domain on the pocket computer were drawn. The user interfaces of the software developed on the pocket computer were shown below in figures between 10 and 13.
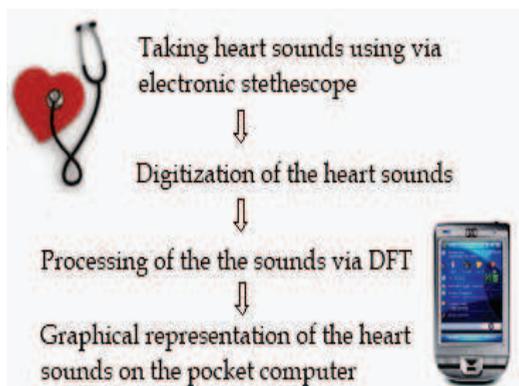

Fig. 9. Flow Diagram of the System

On the introduction screen of the software produced with C# programming language in Microsoft Visual Studio 2005 medium, there appears a selection screen shown in figure 10 on which you could select one of the heart sounds that was at first taken via Littmann 4100 model electronic stethoscope and converted into *wav* format on the pocket computer.

On this screen, the heart sound that belongs to a patient is selected, and the selected heart sound is prepared with the algorithm adapted to the pocket computer from *wav* format to the signal processing phase. After that, it moves to graphic interface. On the graphic interface, after selecting the button named as "Graph", the software on the computer subjects the sound data that was digitized beforehand to DFT. After this process is completed, the data obtained from the DFT is sent into another array. As the signal processing phase is finished, the graphic of the processed heart sound in the time domain is drawn on the upper screen of the pocket computer (the graphic plane writes as "SND" on

Fig. 10. The selection of patient's heart sound data to be examined

the left), and the frequency spectrum obtained as a result of DFT is drawn on the lower screen of the pocket computer (the graphic plane writes as "DFT" on the left). This process almost takes between 5 and 10 seconds. The screen display obtained by drawing on the pocket computer of the time domain graphic and frequency spectrum that belongs to a normal heart sound has been given in figure 11.
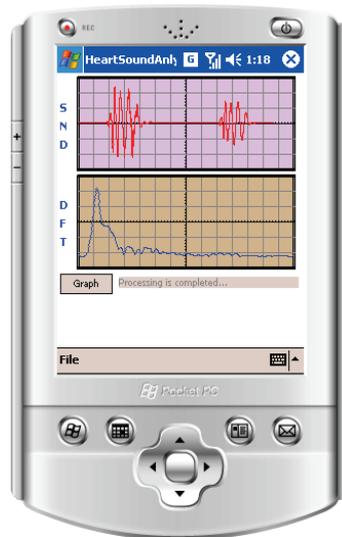

Fig. 11. The screen display with drawing of time domain graphic and frequency spectrum of heart sounds taken from a normal heart

In figure 12 and 13, the screenshots of time domain graphics and frequency spectrum that belong to the heart sounds in which in turn Pulmonary Stenosis and Mitral Stenosis disorders were detected were shown on the pocket computer.
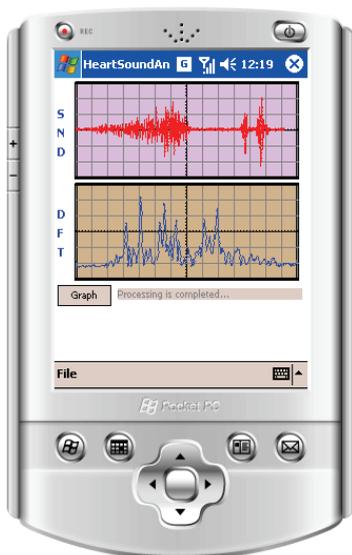
Fig. 12. The screenshot with drawing of a time domain graphic and frequency spectrum of a heart sound that owns Pulmonary Stenosis Heart Disorder
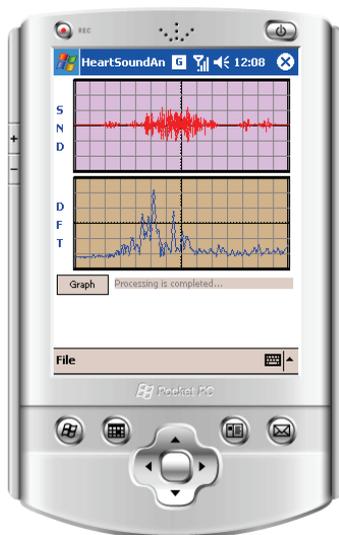
Fig. 13. The screenshot with drawing of a time domain graphic and frequency spectrum of a heart sound that owns Mitral Stenosis Heart Disorder

When there is a comparison between the frequency spectrums of normal heart sound and the heart sounds that have mitral and pulmonary stenosis heart disorders, it is noticed that the heart sounds taken from the normal heart possess less frequency constituents. (b. Güraksın et al., 2009). Because a healthy heart produces a periodical sound while a heart that has any kind of disorder produces some different sounds other than S1 and S2 sounds. Thus, these sounds include noise. The irregular and turbulent blood flows that cause all these sounds lead to inclosure of high frequency constituents in heart sounds.

5. Conclusion

In this actualized study, the heart sounds gained with the use of electronic stethoscope were digitized and then subjected to DFT. Finally, the graphic in the time domain and frequency spectrum that belong to the heart sounds was obtained on the pocket computer. Thus, frequency spectrum of normal and abnormal heart sounds was gained via DFT. As a result, the physicians were prepared with more data in the course of auscultation. By providing the physicians with alternative methods other than listening, it was aimed to help them make a more accurate and faster diagnosis.

The basic purpose in this study is not to diagnose the heart sounds directly. Instead, it was aimed to form a substructure for the prediction methods that may be used in the diagnosis phase. This formed structure sets a substructure for the artificial intelligence programs such as neural network, support vector machines, neuro-fuzzy, which are used for classification**.** (c. Güraksın et al., 2009). The obtained DFT data can simply be made use of in the classification mechanisms.

6. References

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