A NEW HARDWARE DESIGN FOR CARDIAC PASSIVE ACOUSTIC LOCALIZATION

by

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B.S., Electronics Engineering, Erciyes University, 2002

Submitted to the Institute of Biomedical Engineering
in partial fulfillment of the requirements
for the degree of
Master of Science
in
Biomedical Engineering

Boğaziçi University
July 2005

A NEW HARDWARE DESIGN FOR CARDIAC PASSIVE ACOUSTIC LOCALIZATION

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DATE OF APPROVAL: 22.08.2005

ACKNOWLEDGMENTS

First of all, I would like to thank my thesis supervisor, Assoc. Prof. Dr. H. Özcan Gülçür for his great encouragement, help and support throughout this study.

I would like to express my deep gratidude towards Yıldırım Bahadırlar for sharing his experience with me by giving vital suggestions.

I would also like to thank to my family for their all kind of support. I couldn't complete such a hard and time consuming work without their support.

My special thank goes to Gökhan Işık who encouraged me and helped me handle especially the hardware part of the work.

I also would like to thank to my best friends Mustafa Küçükkatırcı and Serkan Karaca for being with me throughout the thesis.

Finally, I am thankful to all humanitarian people in Biomedical Engineering Institute at Boğaziçi University, especially Erhan Uyanık, Yüksel Yazıcı, Ömer Şaylı, Uzay Emir, Murat Tümer and Mustafa Fidan.

ABSTRACT

A NEW HARDWARE DESIGN FOR CARDIAC PASSIVE ACOUSTIC LOCALIZATION

Heart sounds contain valuable information about the function of the heart; expert clinicians can diagnose many heart disorders by listening to these sounds. One dimensional visual representation of heart sounds called phonocardiograms (PCG) are also used to facilitate the diagnosis. Although, PCG is an inexpensive, non-invasive diagnostic technique, it has been neglected until recently because of its limitations and enormous improvements in other diagnostic techniques such as ultrasonography, CT and MRI. Recently, a significant study on PCG was conducted by Y. Bahadırlar and H. Ö. Gülçür; they developed a system which is composed of a specially designed multi-sensor probe in the form of a planar microphone array, precision amplifiers, filters and A/D converters, interface circuitry, a PC and special software and obtained 2-D and 3-D images of estimates of sound producing sites in the heart. The original system called CARDIOPAL (short for Cardiac Passive Acoustic Localizer) had some limitations mostly arising from relatively limited technology at the time. It had no ECG channel, the multi-sensor probe had coupling problems on non-smooth chests. Moreover it used DMA for data transmission, which made the system device-dependent. In the present thesis, a new, easy-to-use and more compact hardware for CARDIOPAL is developed. The new system (CARDIOPAL II) can work on most of the current operating systems without problems and get data more accurately in order to increase image resolution. An ECG channel is added to the system and ECG signals are acquired simultaneously with the sound signals. The acquired data is transferred to a PC using a high-speed USB 2.0 interface. Moreover, a new flexible design is developed to avoid coupling problem of the array for non-smooth chests. CARDIOPAL II is battery-powered; surface-mount technology was used for the design of all electronic circuitry to make the final system smaller, lighter, and more resistant to electromagnetic interference. The device was tested by acquiring signals coming from two point sources. The localization of these sources was achieved. The device was also tested by obtaining data from real subjects. No quality loss from the corner microphones due to the coupling problem was observed. ECG signals were acquired simultaneously and it was observed that the relationship between ECG and sound signals matched with theory.

Keywords: Phonocardiography, Passive Acoustic Arrays, Source Localization, Electrocardiogram, Multi-Channel Data Acquisition, Universal Serial Bus.

ÖZET

KARDİYAK PASİF AKUSTİK KONUMLANDIRMA İÇİN YENİ BİR DONANIM TASARIMI

Kalp sesleri, kalbin işleyişi ile ilgili önemli bilgiler içerir. Klinik tedavi uzmanları, bu sesleri dinleyerek bir çok kalp bozukluğu için tanıda bulunabilirler. Fonokardiyografi (FKG) olarak adlandırılan kalp seslerinin tek boyutlu gösterimi bu teşhisleri kolaylaştırmak için kullanılır. FKG ucuz ve invaziv olmayan bir tanı aracı olmasına rağmen, yararlılıklarının sınırlı oluşu nedeniyle ve Ultrasonografi, CT ve MRI teknolojilerindeki kayda değer ilerlemelerden dolayı son zamanlara kadar ihmal edilmiştir. Son zamanlarda, Y. Bahadırlar ve H. Ö. Gülçür tarafından önemli bir çalışma yapılmıştır. Bu çalışmada, özel tasarlanmış düzlemsel bir mikrofon dizisi, hassas kuvvetlendiriciler, süzgeçler, örneksel/sayısal çeviriciler, arayüz devresi, bir PC ve özel bir yazılımdan oluşan bir sistem geliştirilmiş ve bu sistemle kalpteki ses kaynaklarının kestirimlerini gösteren 2 ve 3 boyutlu imgeler elde edilmiştir. CARDIOPAL olarak adlandırılan ilk sistemin, daha çok geliştirildiği zaman dilimindeki sınırlı teknolojiden kaynaklanan bazı yetersizlikleri vardır. Bir EKG kanalına sahip değildir ve mikrofon dizisi düzgün olmayan yüzeylere tam olarak oturmamaktadır. Buna ek olarak, veri alışverişi için, sistemi yalnızca belirli tip bir donanımda çalışmaya zorlayan DMA protokolünü kullanmaktadır. Bu tezde, CARDIOPAL için yeni, kullanımı kolay ve taşınabilir bir donanım geliştirildi. Yeni sistem (CARDIOPAL II) mevcut birçok işletim sistemi üzerinde sorunsuz çalışabilmekte ve daha yüksek çözünürlüklü imgeler elde edebilmek için daha nitelikli veri yakalayabilmektedir. Sistemde bir EKG kanalı vardır ve EKG işaretlerinin ses işaretleri ile eşzamanlı olarak yakalanması sağlanmıştır. Buna ek olarak, düzgün olmayan göğüslere mikrofon dizisinin tam olarak oturmasını sağlamak için esnek bir mikrofon dizisi geliştirilmiştir. CARDIOPAL II aküden güç almaktadır. Sistemi, küçük, hafif ve elektromanyetik girişimlere karşı daha dayanıklı yapabilmek için bütün elektronik aksamda yüzey montajlı devre teknolojisi kullanılmıştır. Noktasal iki kaynaktan gelen sesler kullanılarak cihaz test edildilmiştir. Sistem bu iki kaynağın yerlerini tespit etmeyi başarmıştır. Cihaz ayrıca, gerçek deneklerden veriler alınarak da test edilmiştir. Köşedeki mikrofonlardan gelen seslerde, mikrofon dizisinin düzgün olmayan göğüslere tam olarak oturamamasından kaynaklanan bir kayıp gözlemlenmemiştir. EKG işaretleri ses işaretleriyle eşzamanlı olarak kaydedilmiş ve bu iki işaret arasındaki ilişkinin kuramda olduğu gibi elde edildiği gözlenmiştir.

Anahtar Sözcükler: Fonokardiyografi, Pasif Akustik Diziler, Kaynak Konumlandırma, Elektrokardiyogram, Çok Kanallı Veri Yakalama, Evrensel Seri Yolu.

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1. INTRODUCTION

1.1 Background and Motivation

Heart sounds contain valuable information about the function of the heart. Listening to heart sounds usually by a stethoscope is called auscultation. Clinicians can diagnose many heart disorders by auscultation. They listen to heart sounds and try to judge on the state of health of the patient by concentrating mainly on their intensity and frequency contents. However, it is so difficult to give meanings to the sounds by auscultation that it needs expertness on this issue. Phonocardiography (PCG), graphic representation of heart sounds recorded over the chest using a transducer, gives the clinicians a one dimensional visual representation of heart sounds. This may facilitate the diagnosis of some heart disorders. On the other hand, the PCG does not have much spatial information. An experienced clinician may only imagine the region where the defect occurs.

Although, PCG is an inexpensive, non-invasive diagnostic technique, it has been neglected until recently because of these limitations and due to enormous improvements in the other diagnostic techniques such as ultrasonography, CT, and MRI. However, application of advanced signal processing techniques to heart sound signals has recently increased the importance of PCG.

Recently, a significant study on PCG was conducted by Yıldırım Bahadırlar and H. Özcan Gülçür [1]. They developed a system called Cardiac Passive Acoustic Localizer (CARDIOPAL for short). CARDIOPAL is composed of a specially designed multi-sensor probe in the form of a planar microphone array, precision amplifiers, filters and A/D converters, interface circuitry, a PC and special software. Heart sounds from different regions of the chest were recorded by the rigid planar array to obtain 2-D and 3-D images of estimates of sound producing sites in the heart. However, CARDIOPAL had some serious limitations mostly arising from relatively limited technology at the time it was developed.

The main objective of the present thesis is to develop a new, easy-to-use and more compact hardware for CARDIOPAL, which can work under most of the current operating systems without problem and get more accurate data in order to have higher resolution images for the future works.

As stated in the previous work, following factors mostly affected the resolution of the images:

1. Signal-to-noise ratio (SNR).
2. Number of the microphones in the sensor array.
3. Closeness of the system model to the true state of the system.
4. Quantization depth.

In this thesis work, two of them, SNR and quantization depth are in the region of interest. The quantization depth was increased from 12-bit to 14-bit in the present design. The entire system is designed in order to keep the signal quality high. A flexible microphone array is realized to overcome the coupling problem of the previous array design. The instrumentation amplifiers are placed right behind the microphones by the help of surface mount technology in order to keep the long cable, which is connecting the amplifiers to the filter stage, from electric and magnetic interference. High-Speed USB 2.0 is preferred for the interface which transfer the digital data to the PC. Thanks to that, the present design enjoys important advantages of USB technology such as speed, versatilely (enabling the device to work under most of the current operating systems without installing any additional drivers). Moreover, the device is designed battery-powered to get rid of any additional protection circuitry and to minimize electromagnetic interference arising from the power lines. An ECG channel is added to the system and the ECG signals are acquired simultaneously with the other channels. CARDIOPAL II is more compact and its power-consumption is lower than the previous one.

1.2 Outline of the Thesis

Chapter 1 introduces the thesis. Chapter 2 gives a brief background of heart sounds and electrocardiogram and explains the previously designed system. In Chapter 3, the new design for CARDIOPAL is given and the test results are discussed. Chapter 4 concludes the thesis and gives suggestions about what is needed to improve the system. In Appendix, schematics of the sound channels and ECG channel are given. The firmware code, the application program code, and the MATLAB code used in this study are given in the CD attached to the back cover of the thesis.

2. HEART SOUNDS AND ELECTROCARDIOGRAM

2.1 Heart Sounds

The heart produces acoustical vibrations during its beating. Listening to the sounds generated by the heart is called auscultation. The sounds generated in one cardiac cycle is divided into four phases. These are called S1, S2, S3 and S4 sounds respectively.

The first heart sound, also called S1, is due to the movement of blood during the contraction of the ventricles. While the ventricle is contracting, the mitral and the tricuspid valves are strictly closed in order to prevent blood from leaking to the atria. The closure of the valves is asynchronous. S1 is further related to to the oscillation of blood between the descending root of aorta and ventricle and vibrations due to blood turbulence at the aortic and pulmonary valves. The reason of the second heart sound (S2) is deceleration of blood in aorta and pulmonary artery and closure of semilunar valves situated between pulmonary artery, aorta and ventricles. The third heart (S3) sound is associated with vibration of muscle walls due to the relaxation of ventricle. Because S2 is a low-frequency and low-amplitude vibration, it is audible only in children and some adults. The fourth heart sound (S4), which is not audible but can be recorded, is related to the contraction of the atria and propelling of blood to the ventricles [2].

Auscultation of heart sounds give significant information to clinicians about heart disorders. Acquiring the heart sounds using a microphone placed on the chest is called phonocardiography (PCG).

Durand and Guardo proposed a novel approach to the genesis of heart sounds.

In their multi degree of freedom theory, the sound producing activities within the heart are due to sudden change of tension in the heart structures and turbulence in blood flow. Because many parts of heart structures are different, their vibration modes are different. Therefore, PCG results from a combination of these modes of resonances and all vibrating structure in the system contribute more or less to the external PCG, depending on their sound energy. Also timing, intensity, and spectral content of the vibrations may vary considerably in accordance with interpersonal and pathological differences, and the overall spectral content at one time is conditioned by the transmission characteristics of the myocardium, a part of lung and thoracic tissue [1].

2.2 Cardiac Passive Acoustic Localization (CARDIOPAL)

The previous design, CARDIOPAL, gives 2-D and 3-D images of estimates of sound producing sites in the heart. This system consisted of a specially designed passive acoustic array, instrumentation hardware and a 2-D array processing software. A subspace based adaptive array processing method called MUSIC algorithm along with a signal model for the sound-source foci on the heart and the propagation of the sounds to the chest was used. Effectiveness of the method was investigated using extensive recording at different SNR levels on phantoms and tested with success on adult subjects as well as on pregnant women. Using this system, different images corresponding to the various distinct phases of the heart beat; e.g., closure of the mitral and tricuspid valves, ejection of the blood in systole, closure of the aortic and pulmonary valves, early and late diastole, were obtained [3].

The system was based on a theory called multi-degree of freedom theory mentioned in the previous section [4].

As stated in the study, the earlier studies show that transmission system from the heart to the chest surface acts as a time-invariant low-pass filter and the intensity of sounds on the chest surface significantly changes between different sites of the chest, although the frequency content does not vary significantly. Considering these findings, fundamental assumptions are made in the design of CARDIOPAL II. These are the followings:

1. The vibrating tissue is small compared with the size of the microphone array and acts as a point source. Therefore, the sound wave from the tissue is propagated as a spherical wave in this very near field.

2. There is no scattering or reflection of the sound wave within the tissue between the heart and the chest surface.

3. The intervening tissue is homogeneous in terms of sound absorption characteristics. The sound velocity is assumed to be constant and can be taken as c = 1530 m/sec.

4. The sound sources are stationary within the observation interval of interest [5].

The multiple signal characterization (MUSIC) method together with the model is used for the localization of assumed point sources in the heart. Locations of the sources are estimated by applying 2-D searches.

Fig. 2.1 shows the maps of images of various phases of the heart beat acquired by CARDIOPAL system.

2.3 Electrocardiogram (ECG)

The electrocardiogram is the graphical representation of the potential difference between two points on a body surface, versus time. The traditional ECG has a

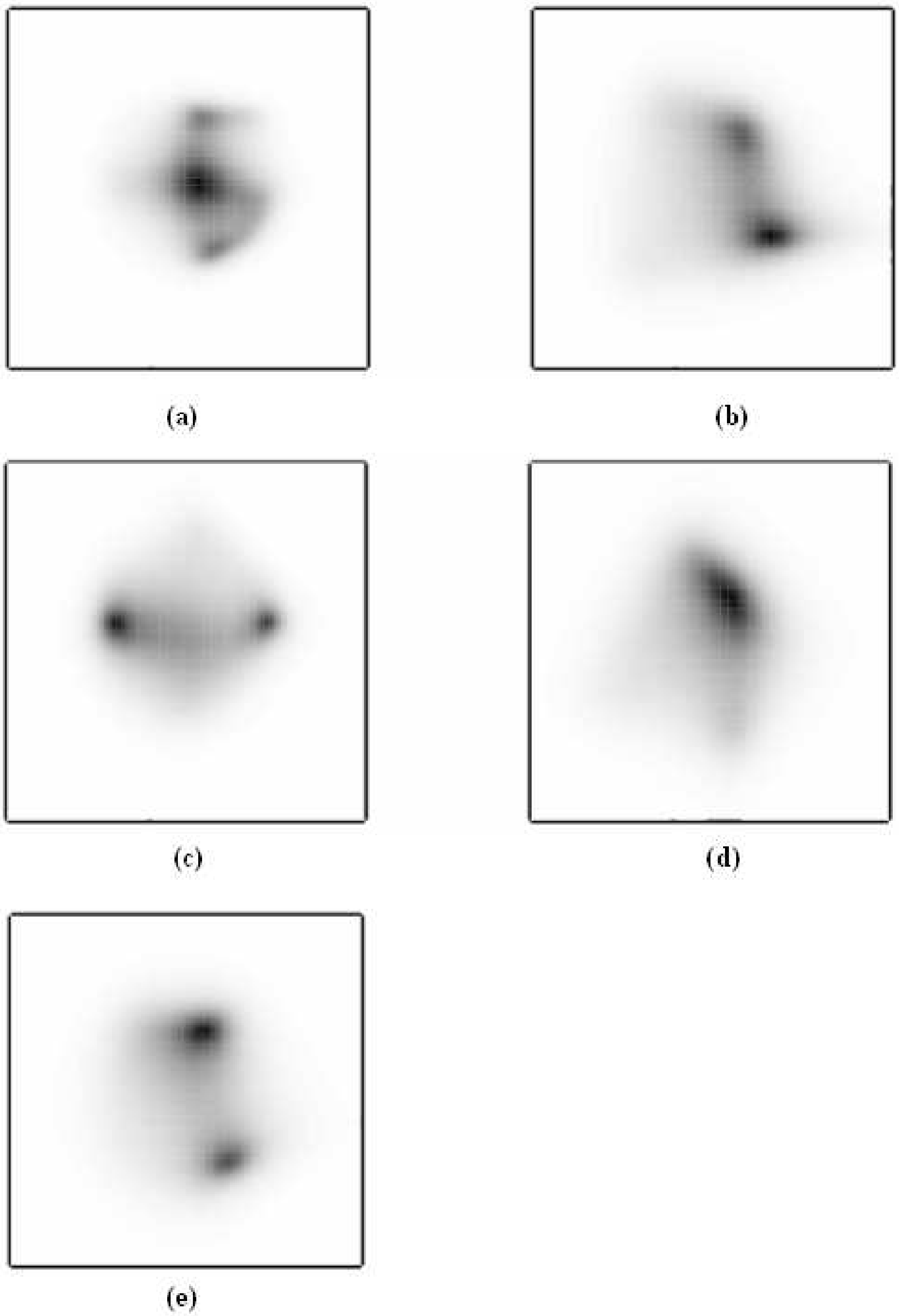

Figure 2.1 The sound localization estimates of a healthy subject and the electrocardiogram (ECG) showing the various phases of the heart beat: (a) the S1, (b) the systole, (c) the S2, (d) early diastole, (e) late diastole.

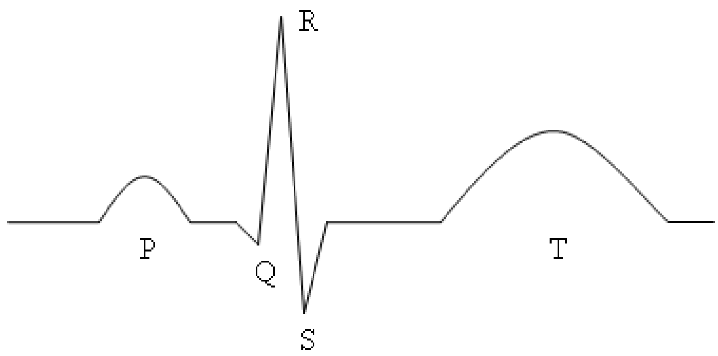

Figure 2.2 A typical ECG waveform

waveform as shown in Fig. 2.2. The deflections have been named P, Q, R, S and T corresponding to atrial contraction (depolarization), ventricular depolarization, and ventricular re-polarization, respectively. The polarity of the signal is entirely dependent on the chosen electrode positions. The QRS complex represents the spread of depolarization through the ventricular muscle, the Q wave being an initial negative deflection, the R-wave the positive deflection and the S-wave a late negative deflection. The normal QRS duration is about 100 ms [6].

There are two aspects to interpretation of ECG. One is concerned with morphology of the “waves” and “complexes” which make up a complete cardiac cycle. The other is concerned with the timing of events and variations in patterns observed over many beats [7].

3. THE NEW DESIGN

The new CARDIOPAL system as shown in Fig. 3.1 consists of a specially designed microphone array, preamplifiers, filter stage, analog-to-digital converters, USB interface and a special software. The system also includes an ECG channel and an ambient noise channel which is identical to other sound channels.

3.1 Microphone Array

Three important points were considered in the design of microphone array:

- Small enough in size to fit an average chest.
- Flexibility.
- Closeness of pre-amplification stage to the array.

In the previous design, the distance between two adjacent microphones in the array was 27.5 mm and the distance between the two end sensors was 82.5 mm. In this work, the distances are reduced to 17 mm and 51 mm, respectively. Although the reduced size of the array affects negatively the quality of the images, this allows the acquisition of subjects who have small chest sizes.

In the previous microphone array design, the surface of the array had a rigid planar structure which prevented the array from coupling properly to the non-uniform human chest wall. Although, the hydrophilic polyurethane matter used to overcome the drawback improved significantly the coupling, the quality of the signals from the microphones at the corner of the array was not the same compared to the signals from the microphones around the middle of the array. In this work, a flexible

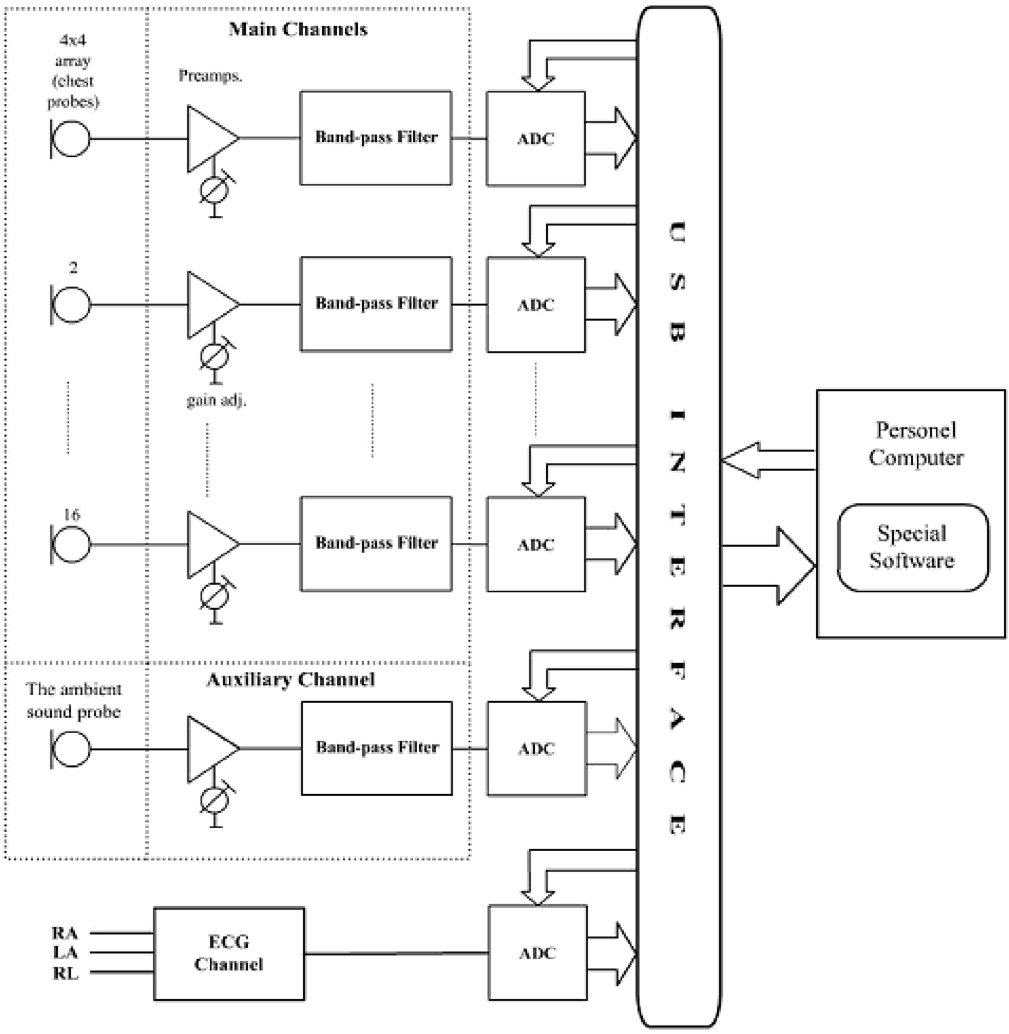

Figure 3.1 Block diagram of the entire system

array is designed in order to overcome this drawback. Each independent microphone and the attached collimation part can move perpendicularly to the surface when a slight force is applied on the chest. This gives the array flexibility so that it fits properly most kind of chests and improves the quality of the signals coming from the corner microphones.

Hydrophilic polyurethane matter is also placed in front of the plexiglas guide. Its effectiveness on signal transfer from the skin was shown by recording sound from the pulmonic area of a healthy subject, with and without polyurethane material

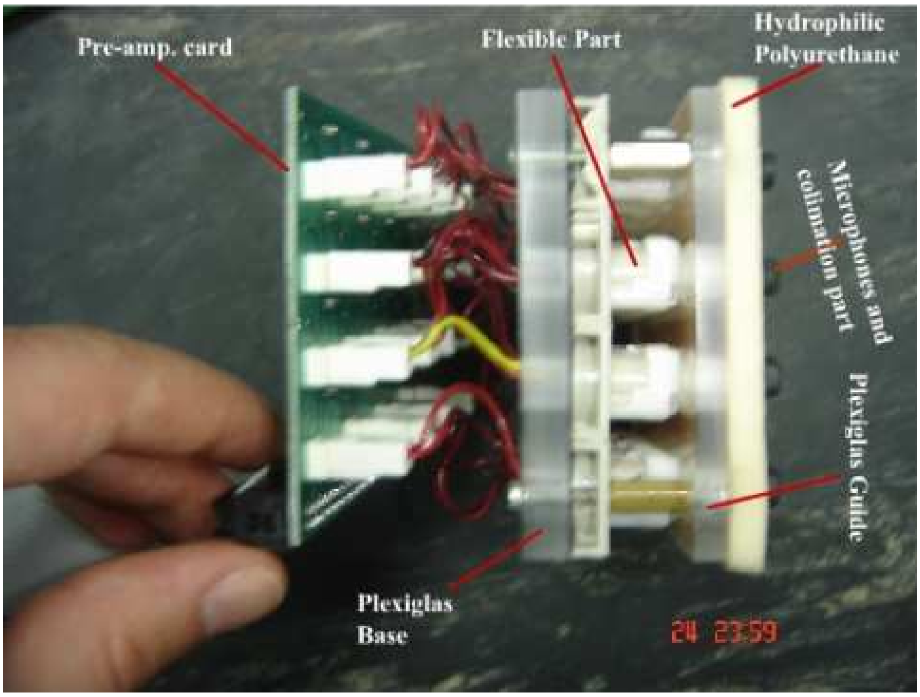

Figure 3.2 Sections of the microphone array

in the previous work. Results showed that it improved the signal transfer to the microphone due to its low loading and porous structure [1].

Air-coupled electret condenser type microphones are used in the microphone array. Although, researches show that less than %1 of the incident sound energy is transmitted across the tissue-air interface due to the impedance mismatch between the chest wall and air [8], there are several advantages of air-coupled electret microphones over the other transducers. First of all, they are very sensitive and the frequency responses are quite good (20-16000 for most type). Secondly, they are much less sensitive to the movement of the body than the other type of transducers because of their air-coupled feature. They are also very small and inexpensive. Because the microphones are omnidirectional, that is they respond evenly to acoustic pressure from all directions, a collimation part is placed on each microphone in order to reduce the angle of acceptance of impinging sound. Because of this, the microphones are more concentrated on the sounds from the chest rather than ambient

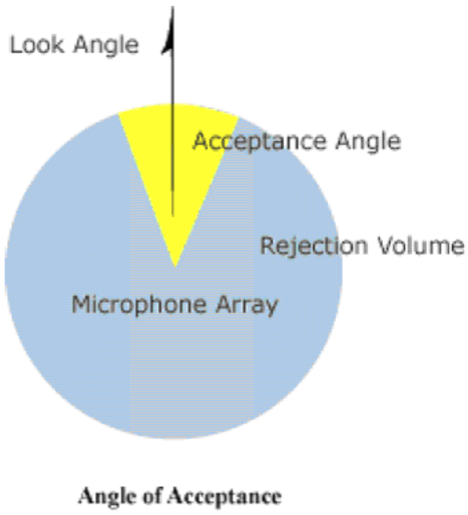

Figure 3.3 Angle of acceptance of the array

noise, which in turn increases the SNR. The angel of acceptance of the array after collimation is shown in Fig. 3.3. In order to measure how the collimator affects the measurement, heart sounds from the tricuspid region were recorded by using a single microphone with and without the collimator. Results show that using collimator increases the signal quality by decreasing ambient noise. Fig. 3.4 shows the measurements.

3.2 Preamplifiers

First order high-pass filters having corner frequencies of 48 Hz are connected to the outputs of microphones in order to get rid of the DC components of the microphone outputs, which may otherwise cause saturation and distort the signals after amplification.

Cables are important because they are the longest part of a system and therefore act as efficient antennas that pick up and/or radiate noise. In the previous

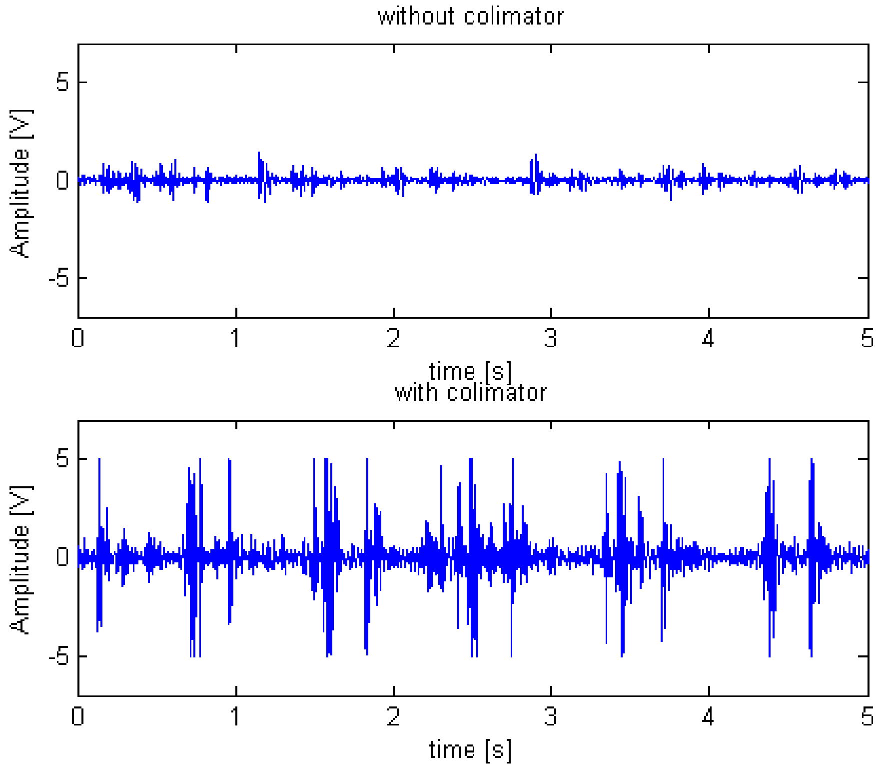

Figure 3.4 The effect of collimation. CH1 is without collimation. CH2 is with collimation

design, pre-amplifiers were far from the microphone array and connected to the remaining part of the system with a long cable without amplification because gathering all pre-amplifiers in a single electronic card needed so much space that it was not practical to place it right behind the microphone array. Although, a twisted pair and shielded cable is used in order to protect the signals from electric and magnetic interference, there was still a possibility for the noise to interfere the raw signals. In this work, with the help of surface-mount technology, a small electronic card which includes all of the 16 pre-amplifiers is designed. Thanks to that, the signals are amplified before connecting to long cables and because the long cables carry high amplitude signals, the possibility for the electric and magnetic noises to reduce SNR to unwanted levels are decreased. This also allows us to use a simple non-shielded flat cable for the connection to the filter stage. The pre-amplifier card is shown in Fig. 3.5.

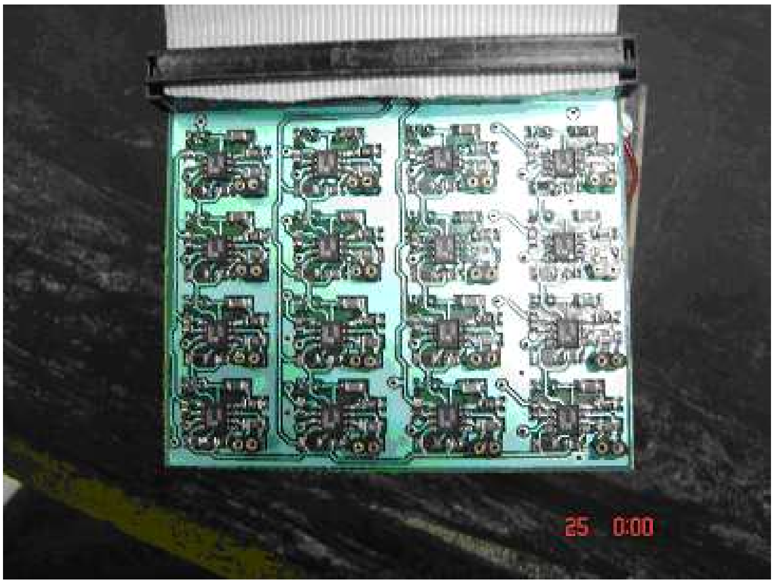

Figure 3.5 Preamplifier card

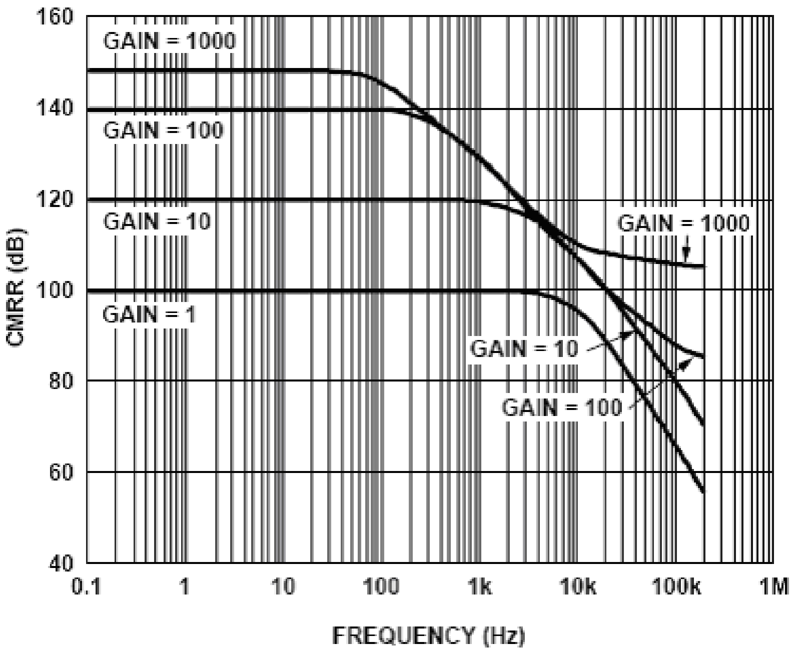

Figure 3.6 Frequency vs. CMRR diagram of AD8221

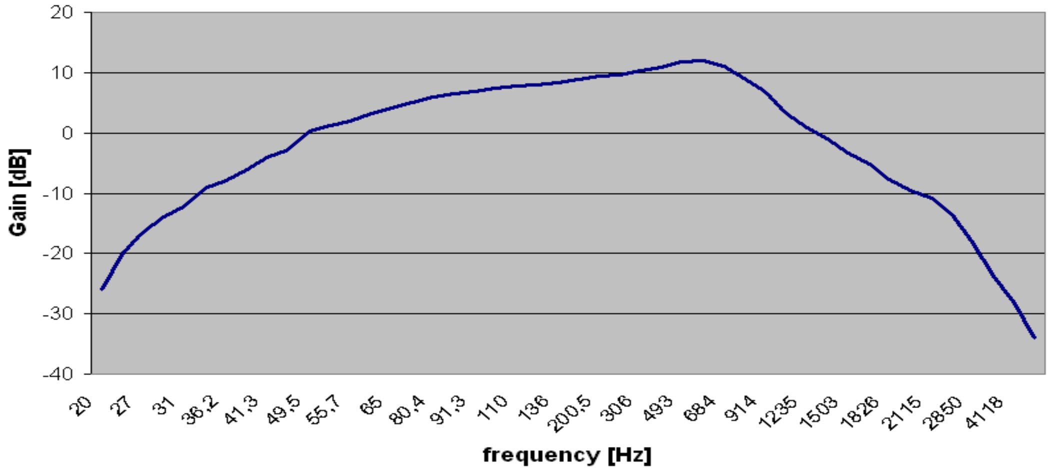

Figure 3.7 Frequency response of the filter stage

AD8221 instrumentation amplifier (IA) in SOIC pocket from Analog Devices was used for the pre-amplification stages. The AD8221 is a gain programmable, high performance instrumentation amplifier that delivers one of the industry's highest CMRR's over frequency. As shown in Fig. 3.6 the CMRR of AD8221 is around 140 dB at a Gain of 100, being higher at higher gains. The IC has also very low input voltage noise (8 nV/$\sqrt{Hz}$ at 1 KHz). The gain for all the pre-amplifiers used in this work is 138.

3.3 Filter Stage

In order to focus on the turbulence related signals and to reduce the power of the low-frequency valvular sounds in the recorded signals, the frequency response is restricted to 80-1000 KHz frequency band [1]. Fourth-order Bessel low-pass and high-pass filters are used to obtain this response. Bessel filter type is preferred because of its linear phase response and constant group delay. Fig. 3.7 shows the measured frequency and Fig. 3.8 shows the phase response of the filter.

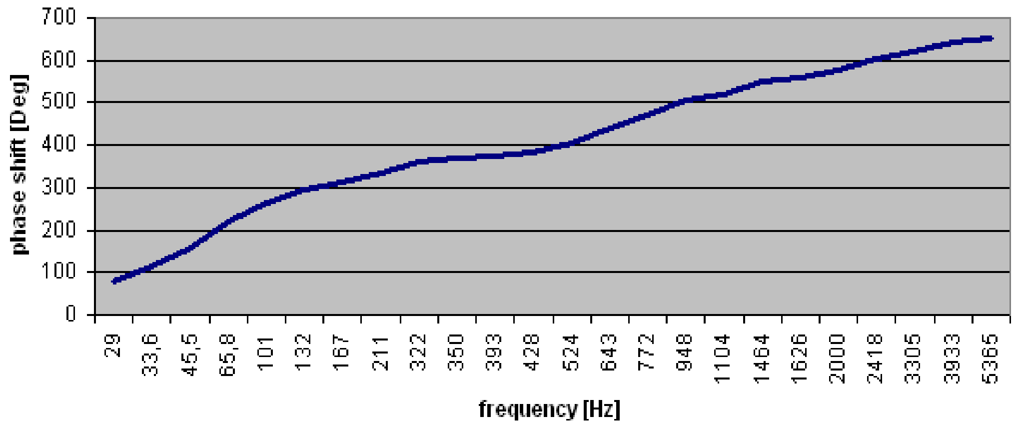

Figure 3.8 Phase response of the filter stage

3.4 Ambient Sound Channel and ECG Channel

The ambient sound channel is identical to the other sound channels. To get best performance from adaptive noise cancelation, the signals acquired by this channel must not be not related to the useful signals coming from the other sound channels but related to the background noise [9]. Therefore, the ambient sound microphone is placed a distance from the chest that ambient noise can be sensed but the heart sounds can not.

The ECG channel is a typical single channel electrocardiograph which includes an instrumentation amplifier (AD8221), a low pass filter, a high pass filter, a notch filter and a right leg driver as shown in the Fig. 3.9.

The electrode/skin interface forms a galvanic half-cell potential. The half-cell potential depends mainly on the condition of the skin and electrodes. This potential varies slowly during measurements, usually because of the movements of the body or electrodes [10]. In order to eliminate these artifacts from the ECG signals, a fourth order Butterworth filter with a cut-off frequency of 0.3 Hz is used. High frequency noise is reduced by a fourth order low-pass Butterworth filter with the the cut-off

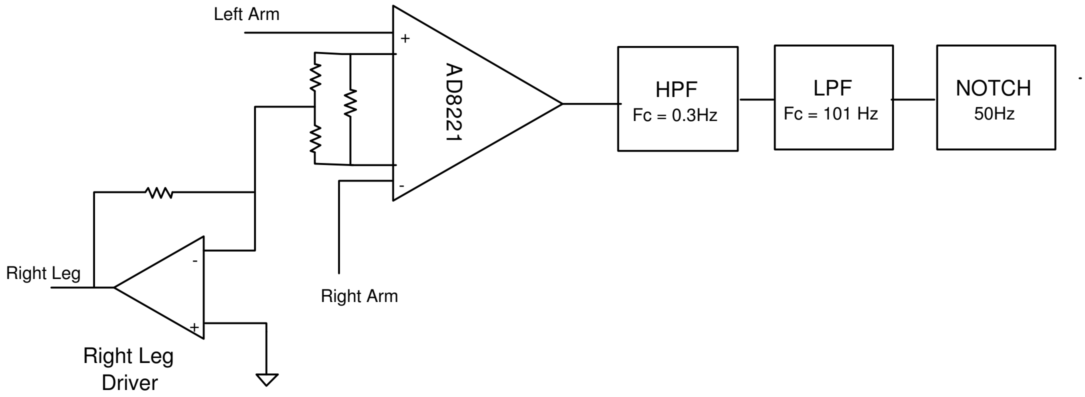

Figure 3.9 ECG circuit

frequency of 101 Hz.

The capacitance between the patient, the power lines and earth cause a small interference current to flow through the body [11]. This common-mode voltage dramatically reduces the quality of measurements. The right-leg driver applies inverted version of the common-mode signal, with the aim of cancelling this interference. Additionally, a notch filter is connected at the output of the band-pass filter to further reduce the 50 Hz noise. Moreover, shielded and twisted pair cables are used to protect the channel from the interferences. The ECG signals are acquired simultaneously with the other channels.

3.5 Analog-to-Digital Converters (ADCs)

The output of the filters are connected to ADC's. The signals are sampled simultaneously and converted to digital form. AD7865 from Analog Devices is used for this purpose. The AD7865 is a fast, low power, four-channel simultaneous sampling 14-bit A/D converter that operates from a single 5 V supply. The part contains a 2.4 μs successive approximation ADC, four track/hold amplifiers, 2.5 V reference, on-chip clock oscillator, signal conditioning circuitry and a high speed parallel inter-

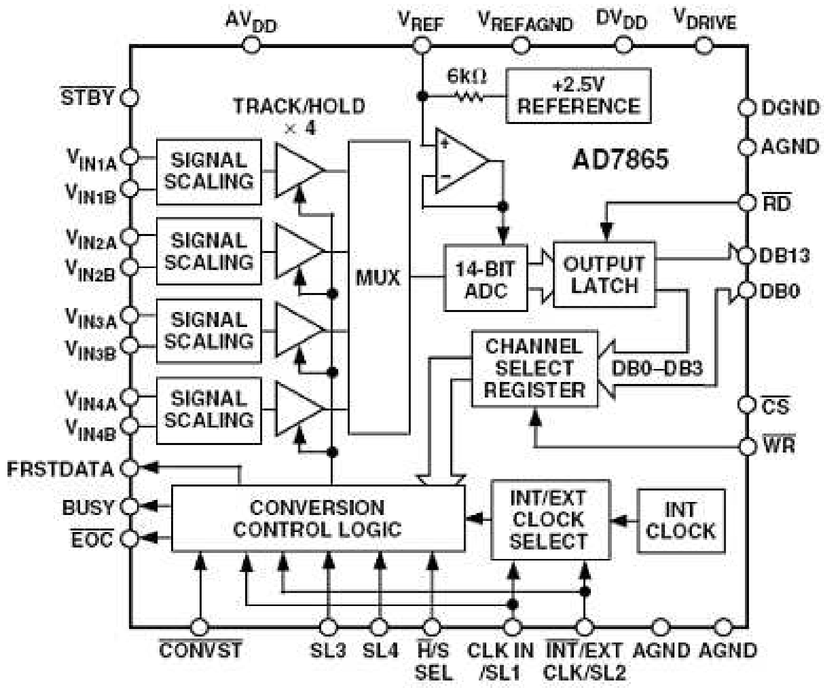

Figure 3.10 Block diagram of AD7865

face. The input signals on four channels are sampled simultaneously thus preserving the relative phase information of the signals on the four analog inputs. The part accepts analog input ranges of 10 V, ±5 V, 2.5 V, 0 V to 2.5 V and 0 V to 5 V. Fig. 3.10 shows the functional block diagram of AD7865 [12].

Because it has four channels, five AD7865 chips are used for the 16 sound channels plus the ambient sound channel and the ECG channel.

In the previous work, the quantization depth was 12-bits. In this work, it is increased to 14-bits. The analog input range of ±5V is preferred in order to maximize the resolution of the ADCs. That is, the full scale (FS) range is divided into 2^{14} or 16384, which means that 1 LSB in 5V FS is only 305 μV. Due to this, low-amplitude and low-frequency sound signals coming from the microphones in the array can be differentiated more easily by the source localization algorithms.

3.6 USB Interface and Data Transmission

One of the main differences between the current work and the previous work is the interface that transfers the data between the data acquisition system and the computer. In the previous design, the data transfer between the device and computer was realized by an interface on the “ISA bus”. In this work, this was achieved using the USB interface. There are many advantages of USB over ISA bus when users and developers are taken into consideration. It is easy to use; when a user connects a USB peripheral to a powered system, windows automatically detects the peripheral and loads appropriate software driver. There is no need to locate and run a setup program or restart the system before using the peripheral. It is faster than most other interfaces; a full-speed interface communicates at 12 Megabits per second. USB 2.0 specification allows communication at 480 Mbps. It is reliable; the hardware specifications for USB drivers, receivers and cables eliminate most noise that could otherwise cause data errors. It is low cost; even though USB is more complex than earlier interfaces, its components and cables are inexpensive. It has low power consumption and is flexible [13][14].

The USB interface implemented in this thesis includes all of the following:

- Cypress 68013 EZ-USB FX2 controller on CY3681 FX2 development board.
- Code in the controller to carry out the USB communication.
- Hardware and code the peripheral needs to control the ADCs.
- A host that support USB.
- Driver software on the host to communicate with the peripheral.
- An application software to enable users to access the peripheral.

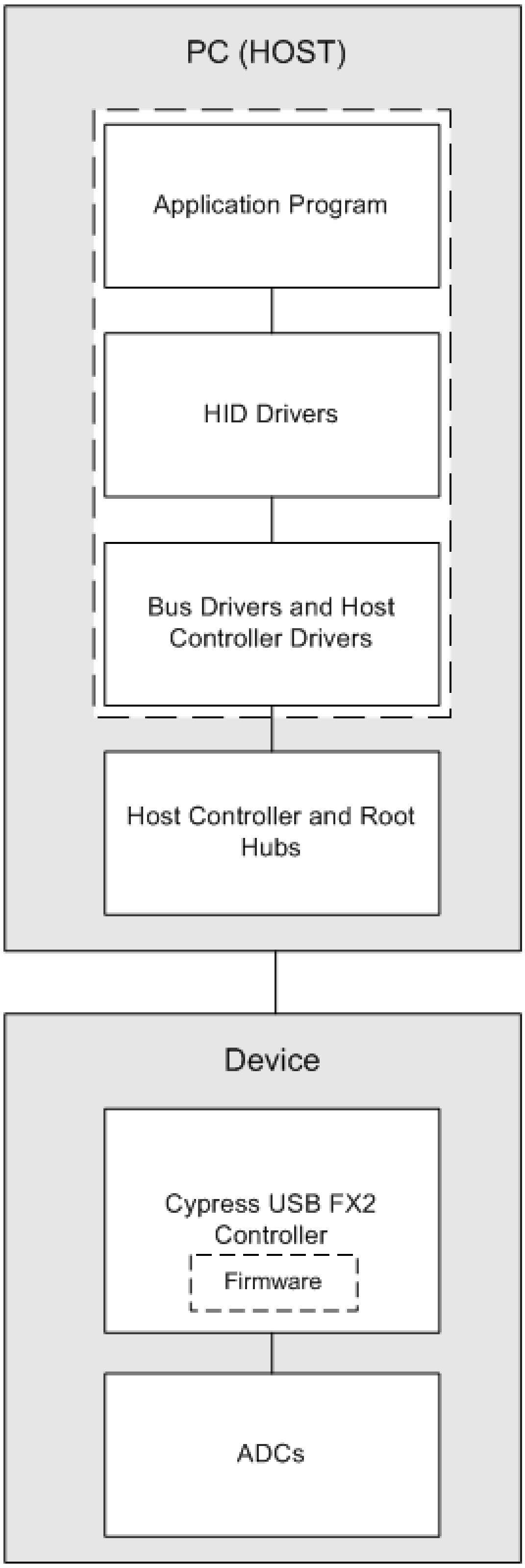

Figure 3.11 USB interface of CARDIOPAL II

The block diagram of the entire USB interface is shown in Fig. 3.11. While the blocks in the dotted lines represent the hardware, the others represent the software part of the interface.

Cypress FX2 packs all the intelligence required by a USB peripheral interface into a compact integrated circuit. As Fig. 3.12 illustrates, an integrated USB transceiver connects to the USB bus pins D+ and D-. A Serial Interface Engine (SIE) decodes and encodes the serial data and performs error correction, bit stuffing, and the other signaling-level tasks required by the USB. Ultimately, the SIE transfers parallel data to and from the USB interface. The CPU is an enhanced 8051 with fast execution time and added features. It uses internal RAM for program and data storage [15].

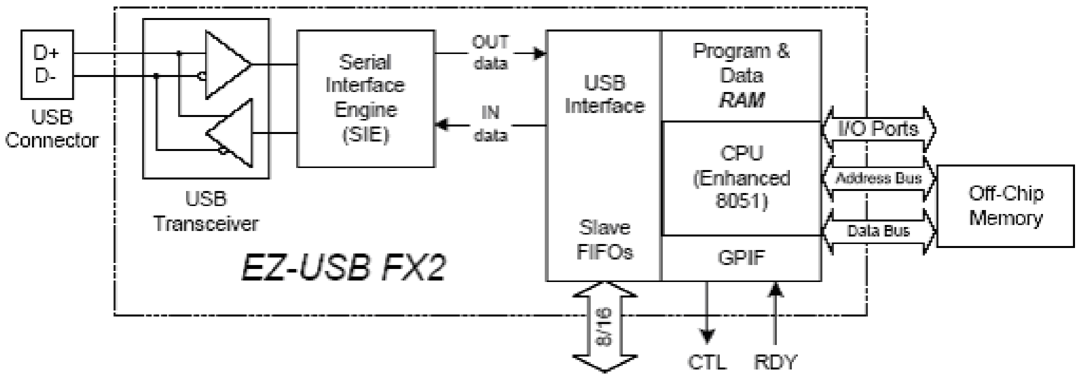

Figure 3.12 Block diagram of FX2

All transmission travel to or from a device endpoint. The endpoint is a buffer that stores multiple bytes. Typically it is a block of data memory or a register in the controller chip. The data stored at an endpoint may be received data, or data waiting to transmit. The host has buffers for received data and data ready to transmit, but the host doesn't have endpoints. Instead, it serves as a starting point for communicating with the device endpoints.

Before applications can communicate with the device, the host needs to learn about what transfer type and endpoint the device supports. The host also must assign an address to the device. The host accomplish this in an exchange of information called "enumeration". FX2 controller can be programmed directly from the computer without needing an EEPROM which includes firmware. The FX2 accomplish this by a two step download called "re-numeration". This means that the entire firmware in the controller can be modified or completely changed without any physical intervention to the device. For example, the transfer rate can be reduced to get small size data.

There are two functions of the firmware. One is to control the conversion and reading data sequences from the ADC's and the other is to establish the USB communication with the host. The firmware in this thesis project is written in C language and compiled by μVision Keil Compiler.

The PC is a host that contains two components: A host controller and a root hub. These work together to enable the operating system to communicate with the devices on the bus. The host controller formats the data for transmitting on the bus and translates the received data to a format that operating system components can understand. USB communication is host centric. That is, a device can not begin USB communications on its own. Instead it must wait and respond to a communication from the host.

The outputs of the channels are connected to the analog inputs of ADC's. $\overline{CONVST}$ and $\overline{RD}$ inputs are common for all ADC's. $\overline{CONVST}$ starts the conversion process. A low-to-high transition on this input puts all track/holds into their hold mode and starts conversion on the selected channels. $\overline{RD}$ is "read" input which is used in conjunction with $\overline{CS}$ low to enable the data outputs. $\overline{CS}$ is chip-select input. The device is selected when this input is active. $\overline{CONVST}$, $\overline{RD}$ and $\overline{CS}$ input of ADC's is controlled by the Port A (8-bit) of the USB controller. 14-bit digital output of the ADC's is connected to Port B (8-bit) and Port D (8-bit). The last two MSB input of the Port D is not used and grounded to ensure that they are logic-low.

When the controller activate the $\overline{CONVST}$ pin, all analog information from the 18 channels are sampled simultaneously and converted to digital form. The digital data is stored by the registers of the ADC's. Then, the controller enables one of the five ADC's by activating $\overline{CS}$ pin of the ADC in order to read four latched 14-bit data. After reading them by activating the $\overline{RD}$ pin for four times, the controller enables the other chips and read the data from them one by one. The controller writes all data to its Endpoint 2. When all 18x2 = 36 byte data is read, the controller gives a new $\overline{CONVST}$ pulse. This process continues for 14 times until there is no place to write in Endpoint 2. Because the capacity of Endpoint 2 is 512 byte, the size of data in Endpoint 2 reaches 504 byte before the host take it. The time between two adjacent $\overline{CONVST}$ pulse is 60,016 μs. That is, the sampling frequency is 16621 Hz.

When the data is filled with 504 bytes of data, a flag bit which tells the host that it is ready to be transmitted is set by the firmware. The host polls the device in fixed intervals specified by the firmware. If the host detects that the endpoint is ready for the transmission during one of its polls, the transmission occurs. Fig. 3.13 shows the data and control connections between the ADC's and controller. Fig. 3.14 shows the flowchart of the data transmission algorithm.

The application program, written in Visual C++, does not know the details of USB communication. It only receives data using the standard Application Program Interface (API) functions. API functions enable device driver and the application program to communicate with each other. The other function of the Application program is to arrange the data acquired from the controller in a mixed form.

Each USB device attached to the host must have a device driver, which is a

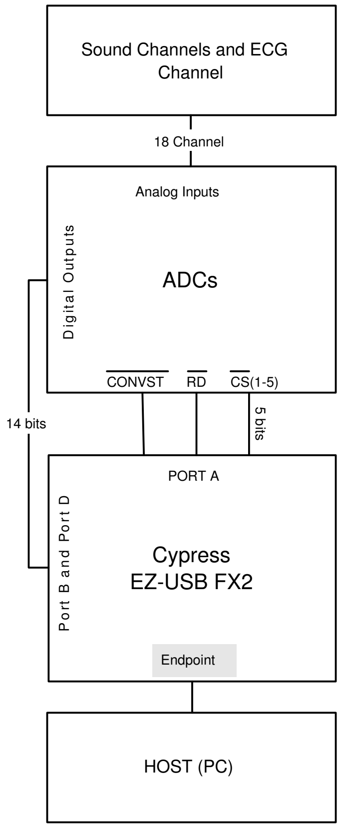

Figure 3.13 Data and control connections between FX2 controller and ADC's

software component that enables applications to communicate with the device. In this thesis work, Human Interface Device (HID) class drivers included in Windows XP, 98 and Windows 2000 operating systems is used. Three main advantages of HID class for the developers and users are in the following:

- It is easy to use; the firmware requirements to classify a device as HID are minimal, consisting mainly of a series of data structures that describe the HID interface and the data to be exchanged.
- Windows 98, 2000, and XP include HID-class drivers, there's no need to write a device driver [16].
- They are currently included in most of the popular operating systems, there is no need to install any driver when users plug-in the device to the PC.

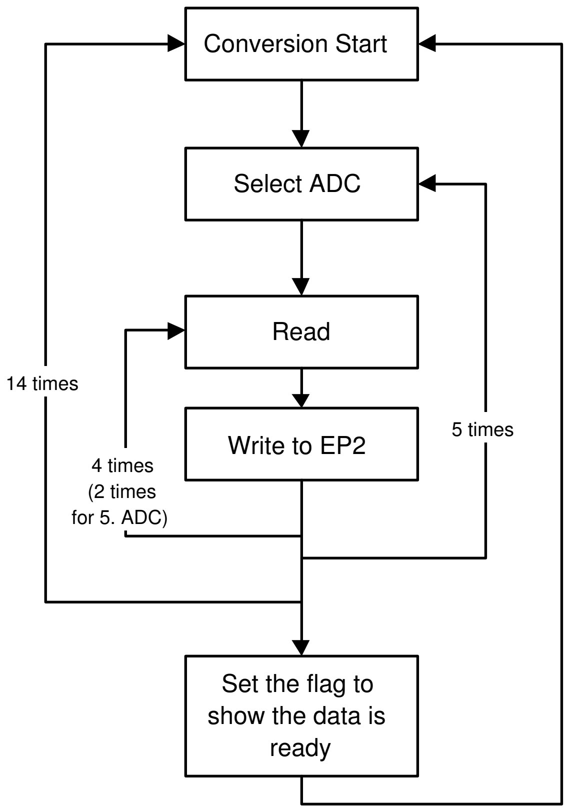

Figure 3.14 Flowchart of data transmission

Because the maximum transfer rate is 12 Mbit/s for full-speed when HID class drivers are used, high speed USB is preferred for the data transmission in order to reach the frequency higher enough than Nyquist frequency of 2 KHz.

Fig. 3.15 shows the instrumentation that includes microphone array, instrumentation amplifiers, filters, ADC's, Cypress EZ-USB FX2 developer kit and PC. Fig. 3.16 shows the complete instrumentation with its housing.

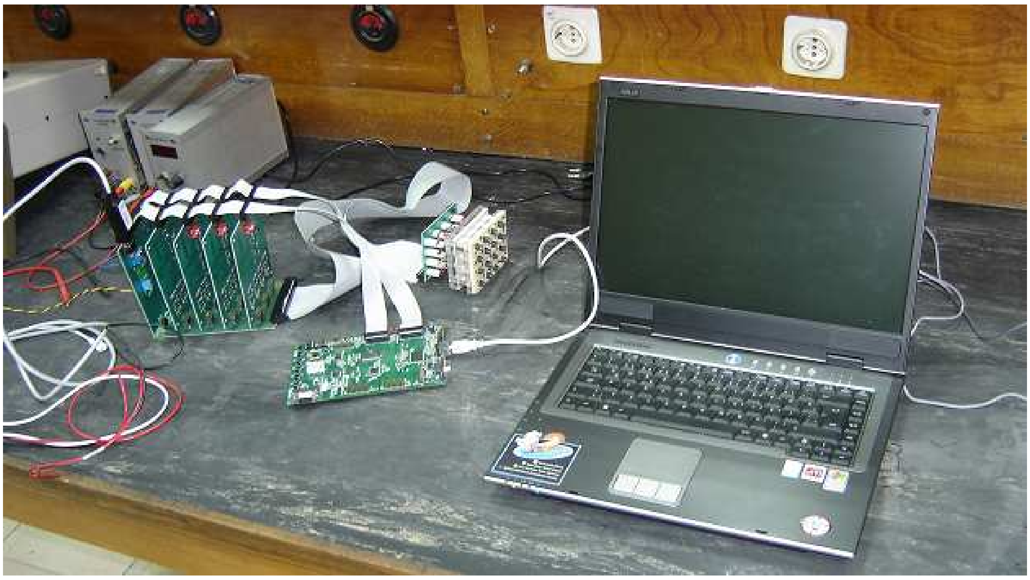

Figure 3.15 Instrumentation that includes microphone array, instrumentation amplifiers, filters, ADC's, Cypress EZ-USB FX2 developer kit and PC

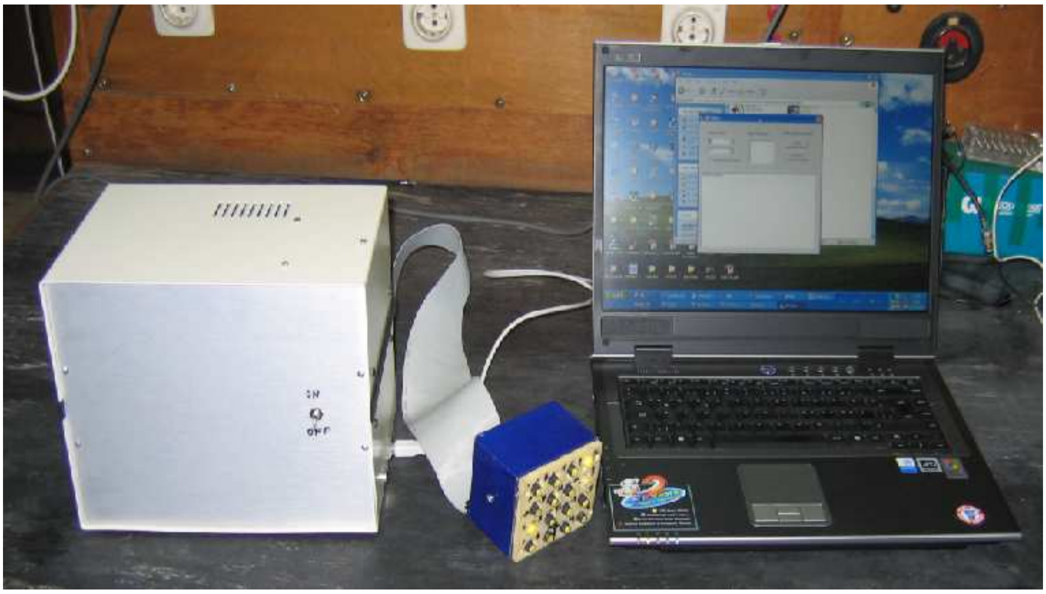

Figure 3.16 Entire instrumentation with its housing

3.7 Hardware Design Features

3.7.1 Crosstalk

Crosstalk occurs by mutual capacitance and inductance by parallel cables and traces in a PCB. One trace or cable (source) induces a certain percentage of its RF voltage into other cable or trace (victim). As the trace approach each other, a larger level of crosstalk is generated [17].

In the design, the long cable connecting the preamplifier stage to the filter stage is specially designed in order to avoid the crosstalk between channels. As seen in the Fig. 3.17 the cable of channel is kept as far as possible from each other. In order to achieve this, two ground wire are placed between two adjacent sound signal wires.

The 3-W rule was taken into consideration to minimize the coupling between traces and signals in electronic cards. The 3-W rule states that "the distance separation between traces must be three times the width of the trace as measured from centerline to centerline of the adjacent traces" [17].

3.7.2 Power Management

The entire hardware of CARDIOPAL II is battery-powered. There are three main advantages of using battery to power the system.

- A small battery-powered device has very small capacitance to the environment, resulting in a low common-mode voltage.
- It improves safety because there are no high voltages present in the amplifier.

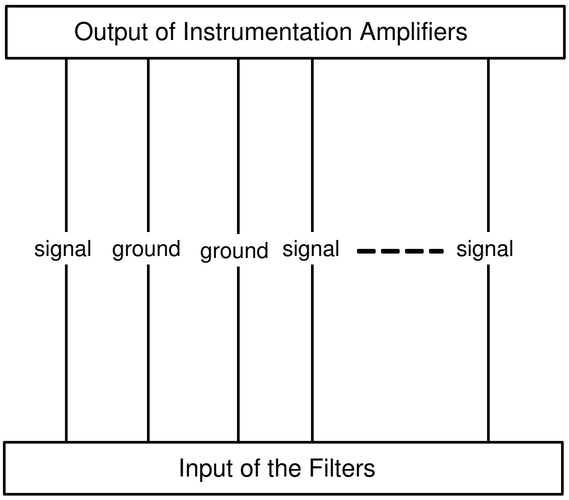

Figure 3.17 Array to device cable design

- Batteries deliver a very clean supply voltage, which is essential for low-noise operation [10].

The present device draws only 130.4 mA from the source. This means that the batteries can power the system for about 10 hours because the capacities of batteries are 2.5 Ah.

Another advantages of using battery is that it simplifies the hardware and reduces the design time because there are no need for the isolation stages.

3.7.3 Surface Mount Technology (SMT)

Surface mount components were used in all boards of the device. The surface components added the following features to hardware:

- The device is smaller and lighter.
- The through-hole component leads serve as tiny antennas that radiate and

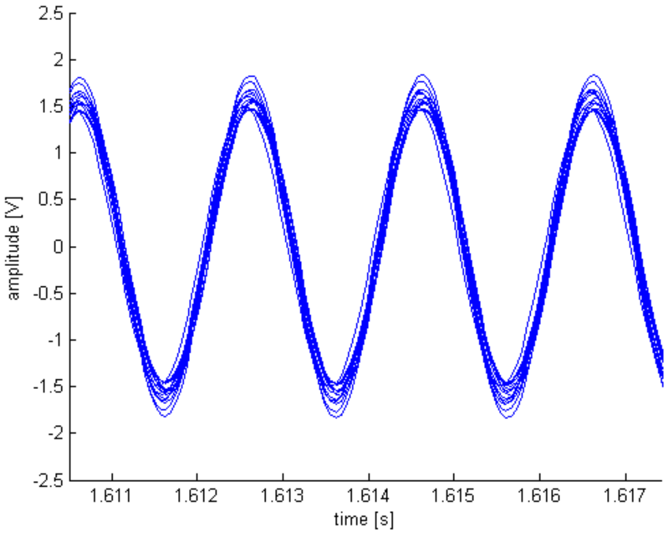

Figure 3.18 Recorded farfield 500 Hz signal after calibration

receive undesirable signals. In SMT, leads do not penetrate the board and thus reduce this problem.

- The surface mount IC's consume less power than the through-holes. This enables the battery to have longer life-span [18].

3.8 Calibration

Because of small differences especially between the passive components, outputs of the sound channels did not give the same responses for the same input. To calibrate the system, a 500 Hz sound source was used. The source was placed in the farfield so that the planar sound waves reach all of 16 sound channels at the same time and impinge them with the same pressure. After acquisition, signals from all the sound channels were equalized with respect to the 16th channel to get rid of amplitude and phase differences. The resultant values were then used in calibration every time after acquisition. Fig. 3.18 shows the 500 Hz farfield waves recorded after calibration.

3.9 Results and Discussion

In a soundproof room, CARDIOPAL II was tested by acquiring a 500 Hz sinusoidal sound signals emitted from a sound source. The medium was air. During the measurement the microphone array was placed at a certain distance from the source in the near field. The angle of the microphone array plane to the source was about 45 degree. The result of the measurement is shown in Fig. 3.20 As seen in this figure, there is a delay between signals coming from two adjacent microphones due to the angle of the microphone array to the source. This means that the microphones which are closer to the source get the signals earlier than the relatively far ones. That is, each signal is about the shifted version of the adjacent one. Although, the test results can not prove the efficiency of the system on real subjects, mainly because of the difference between the speed of sound in tissue and that in the air, it dramatically emphasize the ability of the system to sense spatial information of the source.

The system was also tested using two point sources. Two buzzers as point sources driven by two separate signal generators at 820 Hz and 810 Hz were held 5 cm away from center of the microphone array. The sources were placed on a flat platform and the distance between them was 3 cm. After 1 second acquisition, the signals from all the channels were digital pass-band filtered by a filter having a pass-band width of 30 Hz and the center frequency of 815 Hz. The MUSIC algorithm written in MATLAB by Y. Bahadırlar was then used to localize these sources [1]. Fig. 3.21 shows the localizations of the two sources and the Fig. 3.22 shows pseudo-colored version. The number of the snapshots is 10000 samples, the number of the sources is 5, and exponential loss factor of the medium (ρ) found by trial and error is 4. As shown in the figures clearly, localization of the two near-field sources was successfully achieved.

Experimental data was also acquired from healthy subjects. The microphone array was placed on the fourth intercostal space of the chest and pushed down until the microphone array properly fit the chest. The recording lasted around 5 seconds. The subjects held their breaths during recording to minimize the artifacts arising from lung sounds and chest movements. Fig. 3.19 show the data acquired from one of the subjects. No reduction in the quality of the signals recorded by the microphones due to uneven surface of the chest at the corners of the array was observed. This shows the ability of the array to sense the sounds coming from the corner microphones properly. That is, the coupling problem was solved.

ECG signals was simultaneously recorded from the same subject with the help of three Ag/AgCl surface electrodes. Fig. B.2 shows the simultaneously acquired heart sounds from one of the channels and ECG signals by the system. As shown in the figure, the relationship between simultaneous ECG and sound signals matches with the theory. This also emphasizes that the instrumentation system is successful on simultaneous acquisition. The ECG data is smoothed in MATLAB.

The capacities of batteries used in the device are 2.5 Ah. The device draws only 130.4 mA current from the batteries. This means that the batteries can power the system for about 20 hours.

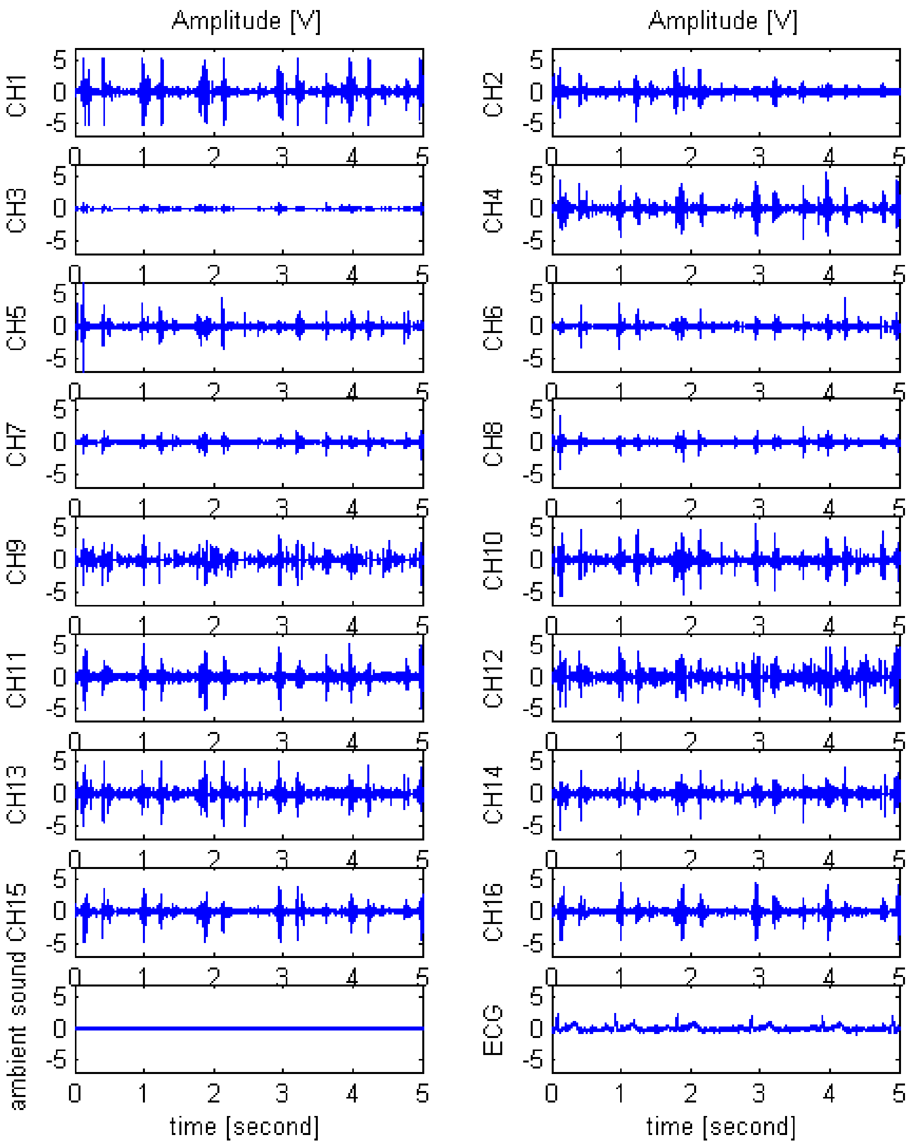

Figure 3.19 A sample data acquired with CARDIOPAL II

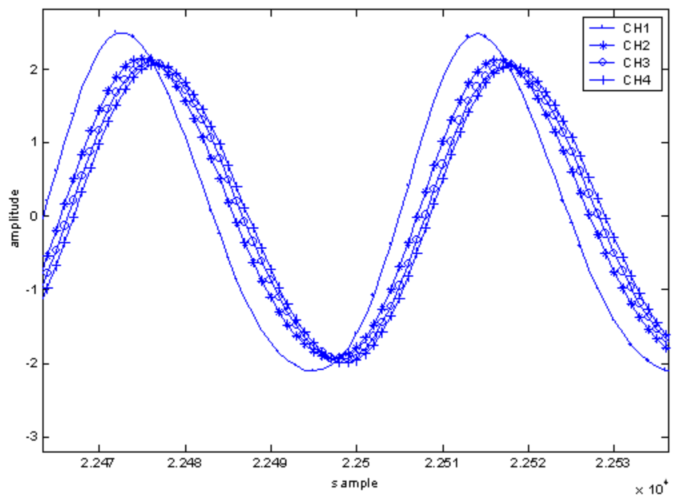

Figure 3.20 Phase difference between detectors observed after recording during the test

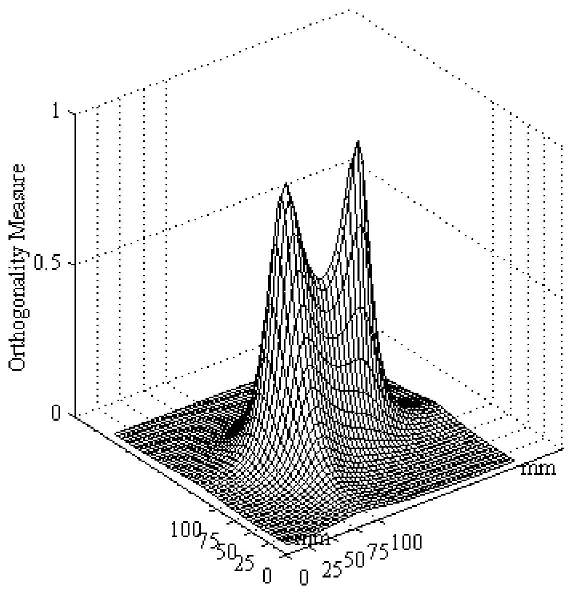

Figure 3.21 Localization of two point sources

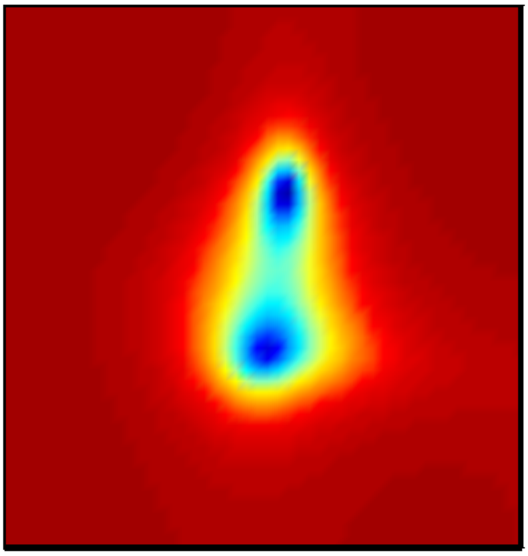

Figure 3.22 Pseudo-colored acoustic map of the point sources

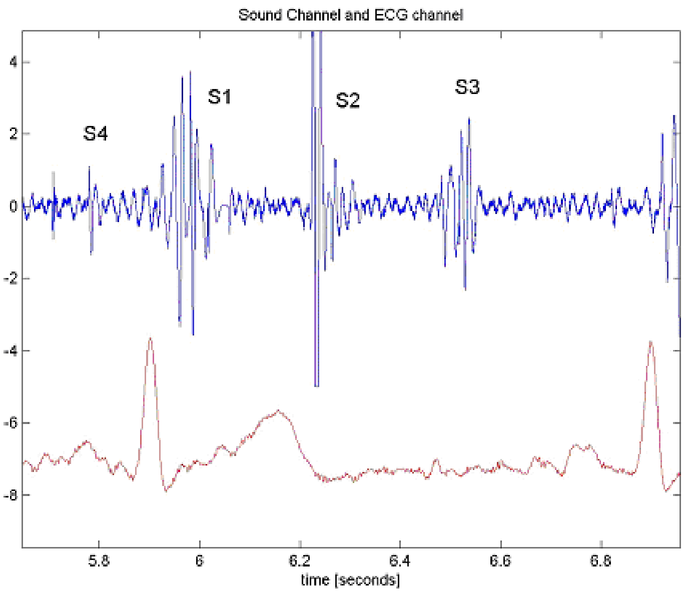

Figure 3.23 Simultaneous recording of ECG with heart sounds

4. CONCLUSIONS AND RECOMMENDATIONS

4.1 Conclusions

A new hardware for CARDIOPAL was implemented. In the design process, especially the SNR and the quantization depth were in the region of interest . Therefore, the entire system was designed in order to keep the signal quality high and the quantization depth was increased from 12-bit to 14-bit in the present design. The instrumentation amplifiers were placed right behind the microphones by the help of surface mount technology in order to keep the long cable, which connects the amplifiers to the filter stage, from electric and magnetic interference. High-Speed USB 2.0 is preferred for the interface which transfer the digital data to the PC. Thanks to that, the present design enjoys important advantages of USB technology such as speed, versatilely (enabling the device to work under most of the current operating systems without installing any additional drivers). The sampling frequency was also increased from 4000 Hz to 16621 Hz.

The results show that the simultaneous data acquisition is working properly. The system can work on all PC's having the popular operating systems thanks to USB technology. Localization of two sources were achieved using MUSIC estimator. The coupling problem of the previous array was solved by a flexible array design. However, the distance between microphones in the array may affect the results negatively. Besides, the flexibility of the array itself may also affect the the localization of images negatively because the location of microphones changes during measurement.

Although, CARDIOPAL II was tested and the the ability of the system to localize two near-field sources was showed, further analysis and experiments are needed to investigate the efficiency of the system especially on real subjects. In order to achieve this, new measurements should be made on a phantom which mimics a real subject.

4.2 Recommendations for Future Studies

First of all, the number of the microphones in the array and the distance between two end sensor should be increased because they directly affect the performance of the system. Secondly, a new acquisition system in which the acquisition process is started by a single button on the hardware should be designed. Moreover, the quantization depth should be increased above 14-bit. Besides, a calibration hardware and software should be developed in order to get rid of the variations in amplitude and phase between channels. Finally, a phantom should be made to test more accurately the system and compare the results with the previous ones.

APPENDIX A. SCHEMATICS OF SOUND AND ECG CHANNELS

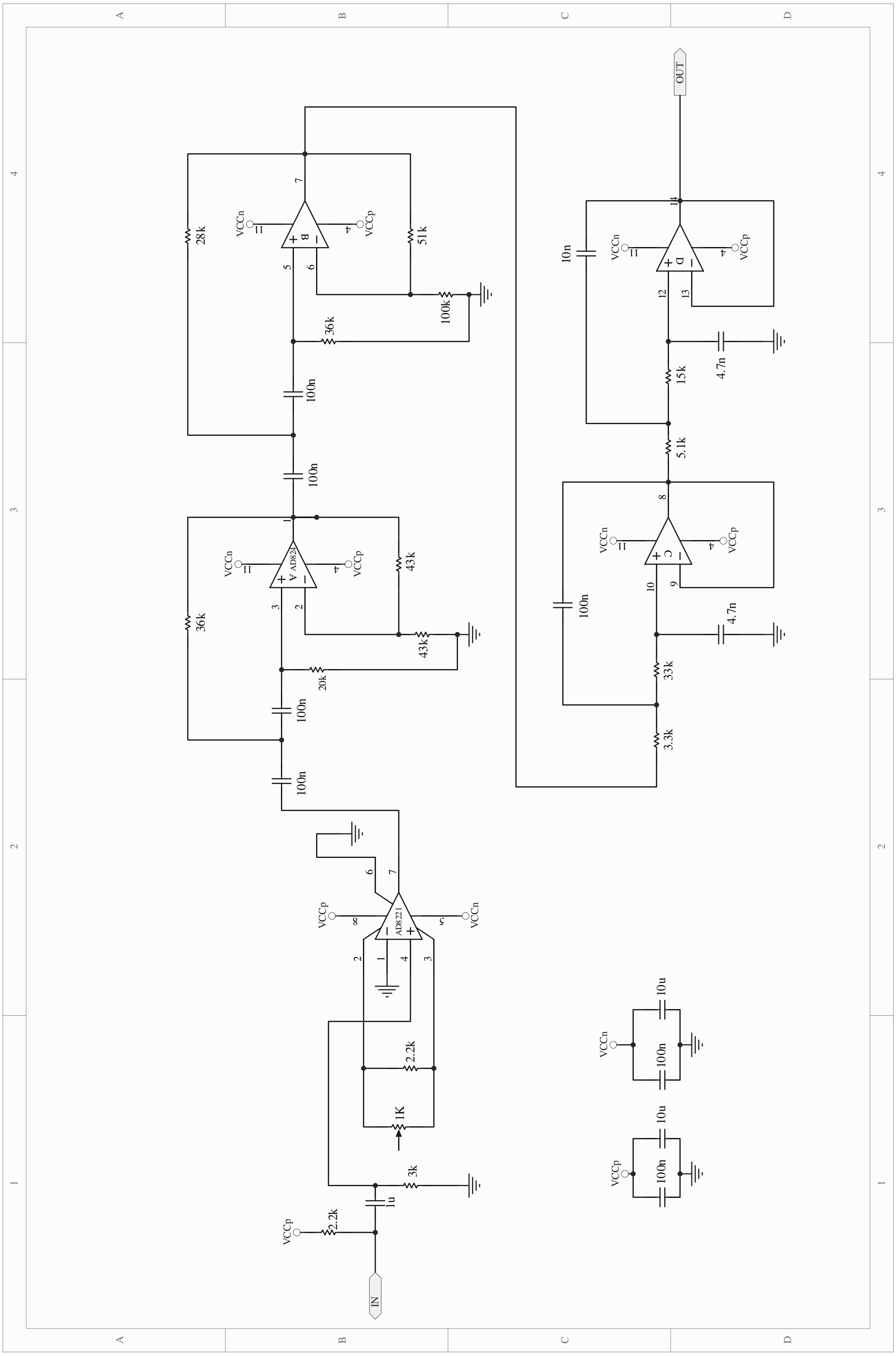

Figure A.1 Schematic of Sound Channel

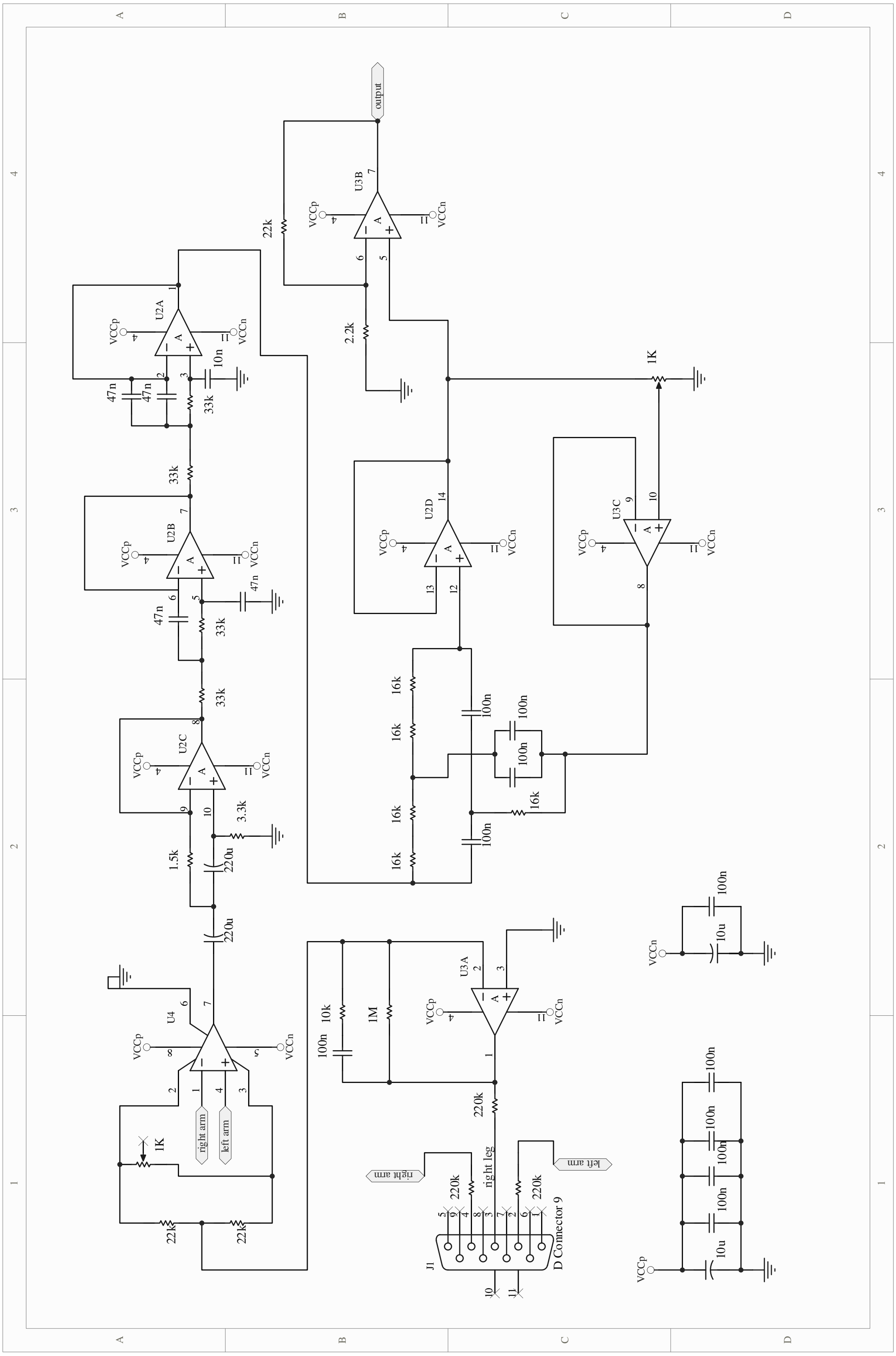

Figure A.2 Schematic of ECG Channel

APPENDIX B. PICTURES OF ELECTRONIC CARDS

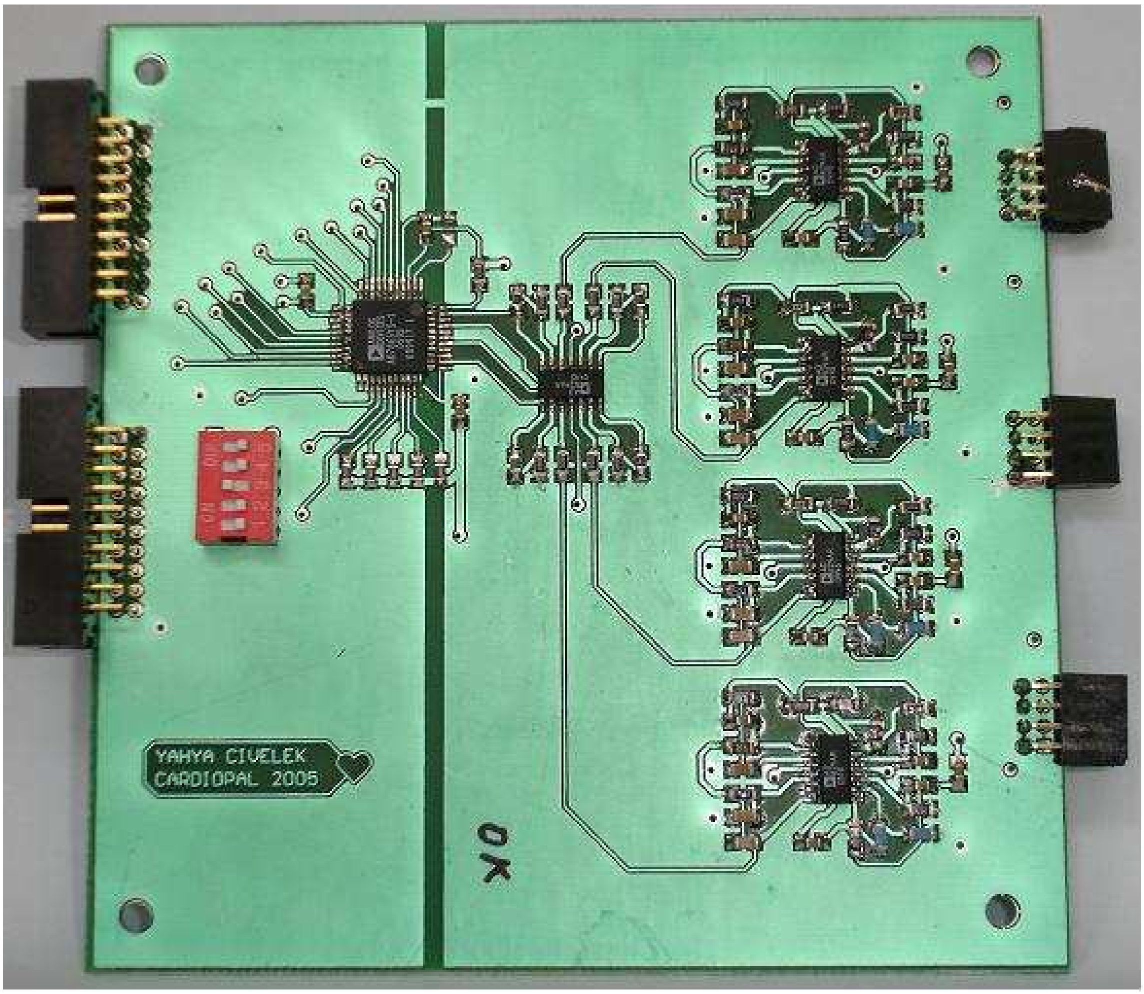

Figure B.1 A picture of the card that includes four filters used in sound channels and the four-channel ADC

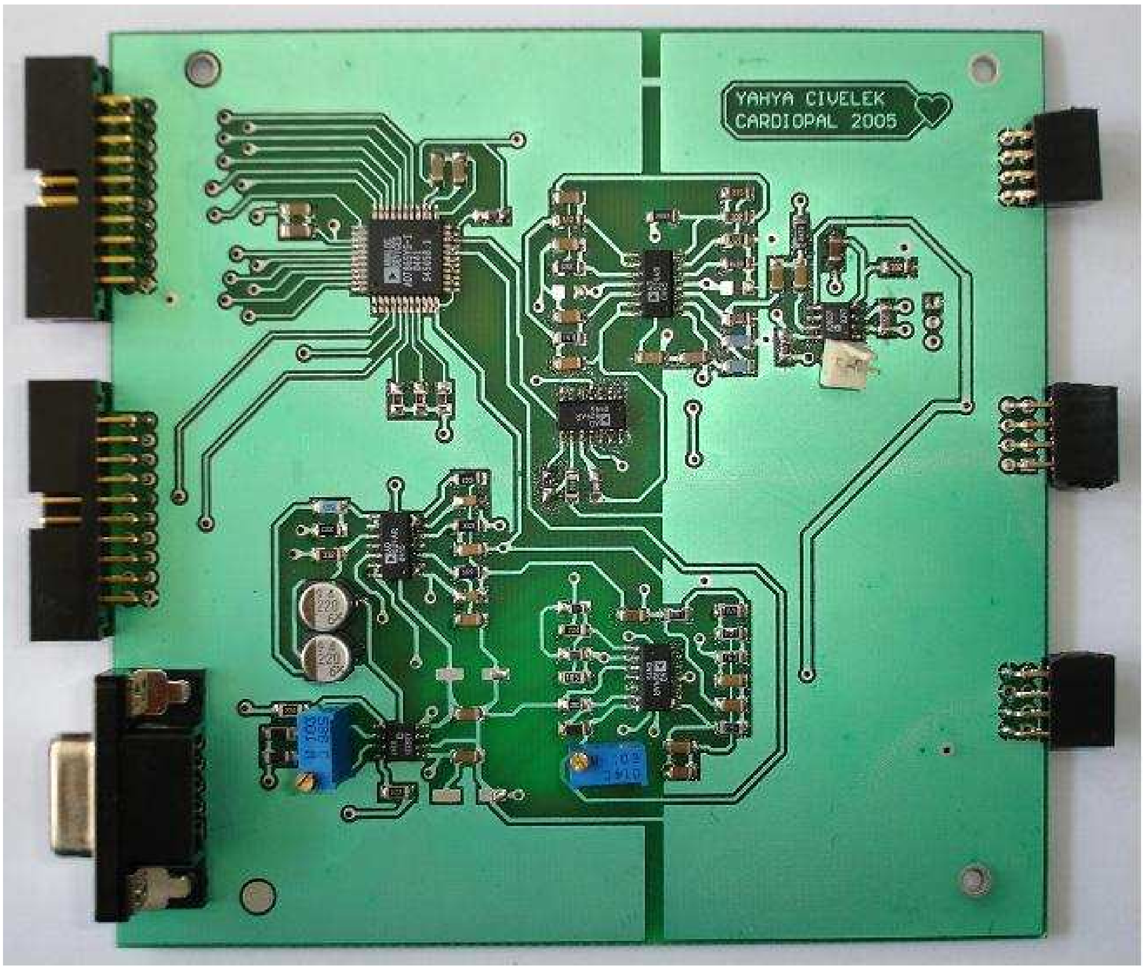

Figure B.2 A picture of the card that includes ambient sound channel, ECG channel and the ADC

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