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Digital Signal Processing System for Stethoscope

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Abstract—This paper presents a design for digital signal processing system for digital stethoscope. The system captures heart beats, filters & analyzes the sound. The result will be displayed using simulation software.

Keywords—Heart sounds, stethoscope, digital signal processing, phonocardiography,digital stethoscope

I. INTRODUCTION

Recognition of heart disease is an important goal in medicine. The majority of stethoscopes currently on the market are acoustic devices that use purely passive mechanical parts to isolate and focus sound generated by the heart. Unfortunately, though these methods have been used for years, the simplicity of such devices is overshadowed by poor sound quality [7]. Some of these devices are highly sophisticated and difficult to maintain. Thus this increases their cost.

The hardware of the system consists of analog and digital parts, respectively. The first one consists of microphone and pre-amplifier. The second one contains a microcontroller with peripherals for data transmission. Simulation software, based on algorithm, will be used for the signal acquisition and digital signal processing (filtering, spectral analysis and others)[4].

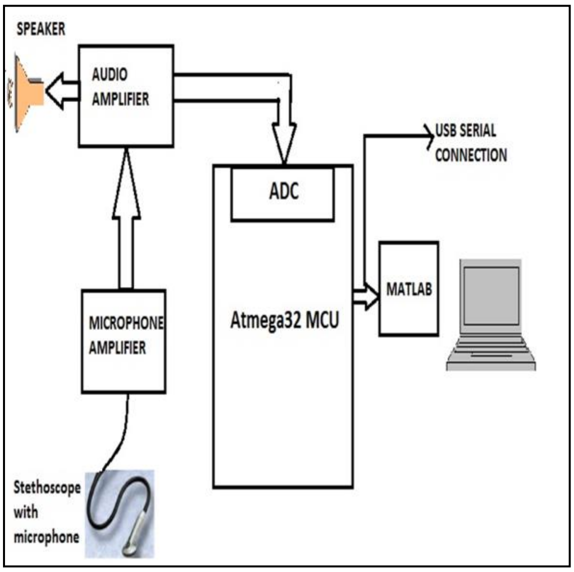

Fig. 1. Block diagram

II. LITERATURE SURVEY

R. Nivethika and N. Kirthika developed designed and implemented a digital stethoscope with heart detection algorithm which they have used MATLAB code for filtering [1]. Thomas R. Harley developed United Active Noise Controlled Stethoscope in which three sensors are used for audio signal, to detect background noise and to remove background noise [2]. Ms. Kadam Patil D.D., Mr. Shastri R.K. designed and developed electronic stethoscope in which they have used zigbee module for wireless transmission [3]. iHeart electronic stethoscope for e-Health & Telemedicine by Andy Su and group uses python language and the result can be obtained using iPhone mobile device via wireless connection [4].

Portable Digital stethoscope was implemented by Dhaval Shah, Harshal Nisar and Satish Phulmali using AtMega32 & analog filtering method [5]. The design & implementation of digital stethoscope for remote exploration was proposed by Trinity College, Hartford, CT, in October 15, 2006 consists of wireless technology to send & receive signal using RF module [8]. 3M Littman electronic stethoscope model 3100 system has ambient noise reduction technology which also uses custom based noise sensor [9].

III. HARDWARE DESCRIPTION

The overall architecture of the system is centered on the ATmega32 microcontroller [1]. The acoustic sensor, microphone amplifier and audio amplifier are inputs to the MCU, while the speaker and software visualization tool are outputs. Figure 1 shows an overview of the system design.

A. Sensor:

The first component of the digital stethoscope system is a sensor, which is able to capture the heart sounds from the places on the chest. The stethoscope acoustic sensor is an integral hardware component of the system. The quality of the sensor directly impacts the quality of the real-time audio output as well as all analyses performed on the measured waveforms. A custom sensor is designed and developed to capture heart signals. The sensor includes a standard stethoscope chest piece to amplify acoustic signals and an electret condenser microphone to convert the amplified signals to electrical waveforms. An electret condenser microphone with a 20 Hz – 20 kHz frequency range is selected in order to capture the entire low frequencies characteristic of internal body sounds.

It is difficult to directly give the output to the ADC, because the signal has low amplitude and it is an acoustic signal, so during the recording is a huge noise from the extrinsic environment [10]. For this reason signal filtering and processing is required.

B. *Microphone Amplifier :*

The microphone in the acoustic sensor needed to be biased in order for proper operation [2]. In addition, the output of the microphone is on the order of millivolts, which is relatively small in magnitude compared to the precision of the ADC sampling the sensor. This makes it challenging for the microcontroller to detect changes in sensor output. In order to address both these issues, a bias and amplifier circuit is designed and implemented to interface the raw sensor output with the MCU. The goal of the circuit is to properly bias the microphone and amplify the sensor output to detect voltage swings caused by sounds.

C. *Audio Amplifier & Low Pass Filter:*

Low pass filter will be implemented in microphone amplifier circuit. Smoothing of signal takes place at this stage [11]. The gain of the circuit is given by the following equation [5].

$$G=1+\frac{R_4}{R_1} \tag{1}$$

In the circuit shown in figure 2 the most important element is an operational amplifier, which should be low noise, have high slew rate and voltage gain. Besides that in usage there are passive elements such like capacitors and resistors to adjust sensitivity and amplification. And hence we use audio amplifier circuit with LM386 Low Voltage Audio Power Amplifier. This audio amplifier circuit gives output to speaker as well as Atmega32.

D. *Analog to Digital Conversion:*

We will use the ADC of the Atmega32 microcontroller [5]. ATmega32 has 8 ADC input channels and each channels give 10-bit resolution. We can use ADC in 8-bit also by setting ADLAR bit in ADMUX register. We are using ADC in 8-bit and auto trigger mode, so ADC will start next conversion on the next clock cycle of last conversion. The ADC value is compared with the threshold value set by user and if it is greater than the threshold value then starts Timer1. Timer1 is used to measure the time period between two peaks, which are used to determine the heart rate. When we get second time peak stop Timer1 and take the value. We send these values to PC through USB serial bus.

IV. DIGITAL SIGNAL PROCESSING

A windowed sampling scheme will be used [1]. This basically means we take a string of samples over a certain time period, process the data, determine the BPM, and start the next sample string. In order to cover the reasonable BPM ranges of the human cardiovascular system, we are aiming for reliable performance between 60 and 180BPM. Our detection scheme basically attempts to find pulse peaks in the data, determines the time between the peaks, and then converts that to BPM [6].

The sampling frequency of 500Hz will be set. Then according to Nyquist Sampling Theorem, the largest frequency component present in our sampled signal should be 250Hz. This in effect is a low pass filter.

V. ANALOG CIRUIT IMPLEMENTATION

We have developed our schematic in Cadsoft Eagle Software. The schematic diagram of our project is as shown in figure 2.

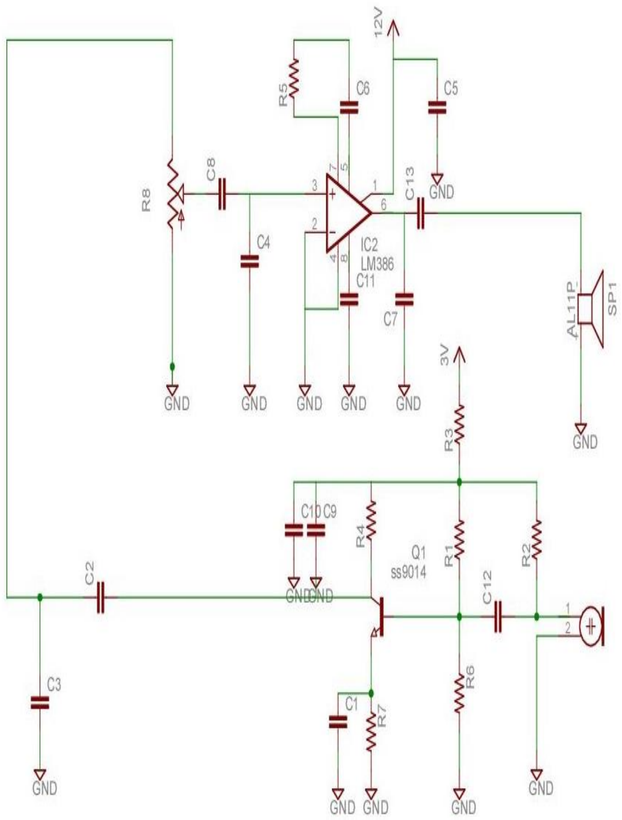

Fig. 2 schematic of analog circuit

Signal from the microphone does not have suitable strength, so it must be amplified in pre-amplifier circuit. We make use of NPN Epitaxial Silicon Transistor ss9014 IC which is a Pre-Amplifier with Low Level & Low Noise characteristics as microphone amplifier.

VI. SOFTWARE DESCRIPTION

A program is being developed to read data from the microcontroller through a serial connection. The received data will be then processed and displayed on a plot in real time. The program is also designed to calculate the beats-per-minute (BPM) of a cardiac waveform in real time as well. An average of a cardiac waveform will be taken to filter out ambient noise and detect when voltage spikes from heartbeats in a clean fashion. The first and second derivative of the average will be required to determine where those steep spikes occur and thus, record the start time for each heartbeat. By calculating the difference between consecutive start times, it will be possible to measure the time for a single heartbeat and to calculate BPM. The program will be coded either in MATLAB or VB.NET based on efficiency, readability of code, debugging and data processing speed .The signal processing algorithm is explained in Fig. 3.

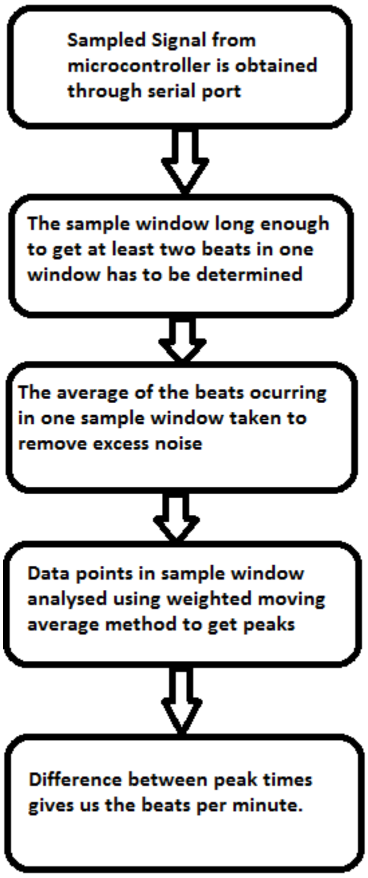

Fig .3 signal processing algorithm

CONCLUSION

This digital stethoscope once developed will be useful for trainee doctors as well as for physician. The system is very easy to handle and maintain. The stethoscope is not bulky hence easily portable.

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