DESIGN AND IMPLMENTATION OF A WIRELESS SURROUND SOUND ELECTRONIC STETHOSCOPE USIGN FPGA PLATFORM

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Abstract- This paper describes a development of a surround sound electronic stethoscope, which split the monaural heart sound signal via transversal filters to produce a pseudo stereo sound for both ears that enables the listener to differentiate the different sounds of the heart easier.

Keywords- Stethoscope, Heart Sound, Surround Sound, Digital Signal Processing.

I. INTRODUCTION

To identify and determine the occurrence of various events by hearing is a natural human instinct. For example, we make corresponding actions after hearing the thunder, baby crying or a door opening. In the clinical medicine application the stethoscope is used for a non-invasive signal detection to listen to the sound of human body, especially generated from the heart and lungs. Through a stethoscope, heart murmur, irregular heartbeat can be found, and by hearing the sound of air flow through the human respiratory tract and lungs, it can also determine whether the relevant organ is abnormal.

The stethoscope uses an acoustic pickup to convey the sound information to both ears of a listener simultaneously via the sealing hose. The acoustic pickup of traditional stethoscope usually has two sides; one side is with a sound film surface and the other side with a bell-shaped face.The sound film surface is mainly used to listen to the high-frequency sounds; it will be pressed gently on the skin surface when used.While the bell-shaped face is for listening to the low frequency sounds, such as the blood pressure measurement, it should be pressed a little harder on the skin when used (**Fig. 1**).



Fig. 1: Construction of traditional stethoscope

When the clinicianuses a traditional stethoscope to heard heart murmur of patients, it is difficult to identify whether it is benign or malignant. The 3M Littmann electronic stethoscope with Bluetooth technology can transfer the heartbeat sound to a computer and display it in real-time waveform, the clinician can then analyze the heartbeat sound waves in detail, and can also listen to the heartbeat sound by deceleration to half for more accurate diagnosis (**Fig. 2**).



Fig. 2: 3M Littmann electronic stethoscopes

Stethoscope can also be used in engineering fields (**Fig. 3**), such as distinguish whether the bearing is damaged by hearing the sounds, the SKF TMST 3 electronic stethoscope can separate critical sound from many background noises of running machine clearly, and the sound can also be recorded



Fig. 3: Industrial electronic stethoscope

for future comparison. This electronic stethoscope can isolate the background noise and provides adjustable sound volume for the user to detect, track

and diagnose the source of noise in all kinds of machine, such as sound of damaged bearing, steam valve, cam noise, piston shock, gears, pumps, and relay or solenoid.

II. MOTIVATIONAND OBJECTS

2.1. Materials and Procedures

As part of a physical examination, preliminary examination relies extensively on cardiac auscultation, the clinician can listen to the sound of the heartbeat, and additional sound or noise, such as blood surging noise for diagnosis. This information is usually transmitted from the patient's chest through one or two hoses of a stethoscope to the medicalexaminer's ear. The shortcoming of traditional stethoscope is the sound through empty long tube to the clinician's ear. Long distance transmission caused a significant loss of the sound intensity with distortion, and it is often affected by some noise. Besides, it has no volume control to enlarge the sound, therefore, the diagnosis result often rely on the clinicians' professional experience. **Fig. 4** shows some examples of heart murmur, the second part of a single heart sound is essential for diagnosis of heart disease, the normal time separation between A2 and P2 is about 40mS.

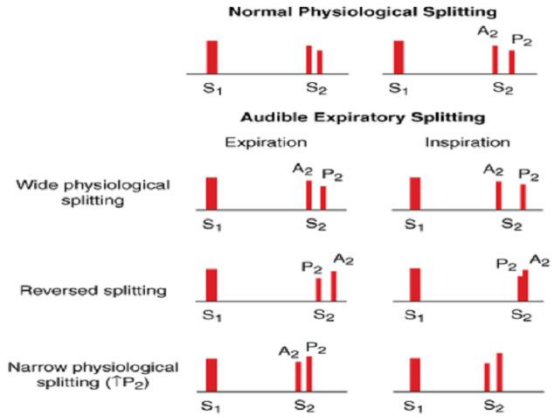


Fig. 4: Examples of different heart sounds

Although the existing electronic stethoscope has overcome some above drawbackseffectively.When listening to the heartbeat, human can identify separate sounds only greater than about 40mS interval, i.e. two separated sounds with less than 40mS interval would be considered as one mono sounds.That could cause a misjudgment, thus there is still rooms for improvement.

The purpose of this project is to provide an electronic stethoscope, which comprises an acoustic pickup head with a contact microphone, an auto gain control circuit (AGC), two transversal filters, a phase split circuit, and an adjustable sound delay line. Additional functions of the developed stethoscope are noise reduction or cancellation, automatic volume adjustment, and the ability to distinguish separate sounds with less than 40mS interval, and improve the clarity of heartbeat sound.

When people use one ear or both ears to listen to the sound at once, two sounds with less than about 40mS intervals cannot be distinguished generally. These very close sounds will be perceived like only one sound, but if the sounds are transmitted to the ears separately, then the different sounds with less than 40mS interval are perceptible to the human brain.

The electronic stethoscope of this project uses this special property of the binaural hearing to listen to the sound. The stethoscope comprises a single microphone for sensing mono acoustic heartbeat and a signal processing circuit for treatment of this signal in the time and the frequency domain to provide different sound signals for left and right ear. By providing the human listener with a coherent sound independently reproduced on each side of the head, any amplitude difference, artificially or naturally generated between the two sides, will tend to shift the perception feel of the sound.This generates a pseudo stereo binaural sound, makes certain heart sounds easier to distinguish for the clinician.

2.2. Working Theory

The electronic stethoscope in this project is a sound synthesis apparatus, which can generate a pseudo stereo sound for two independent audio channels using a controllable time delay lines. The generation and synthesis of the real-time two-channel sounds are implementedthrough a simple circuit. When using this stethoscope to listen to the patient's heart sounds, the pseudo stereo effect started immediately and produces a widened and definitely clear sound in sense. The separation of heart sounds generates a unique effect, in that the clinician perceives a spatial orientation to the sound as if the heart sounds come from within his own head.

Fig.5 shows a method to split the originalheart sounds for the left and right ear. **Fig. 5a**shows a single heartbeat waveform about 1/3 seconds.The heartbeat sound is composed of two normal sounds, which contains low frequency and high frequency components individually. Single heartbeat sound shows in the figure can be divided into "Lub-Dub" two parts, "Lub" is the first heart sound S1, and the "Dub" is the second heart sound S2.

The first sound S1 occurred at the time of closure of the atrioventricular valves of the heart, the higher pitch second sound S2 occurs at the time of closure of the aortic and pulmonic valves. The mixed high and low frequency heart sounds in the original diagram a) can be separated into diagrams b) and c) thatis heard by the medical examiner's ear.The net effect on the listener is the perception of a clear and distinct separation of each heartbeat sound.The listener will think this sounds auscultation is happening in the head. This diagram explains why use the surround sound can better distinct heart sounds than the traditional method.

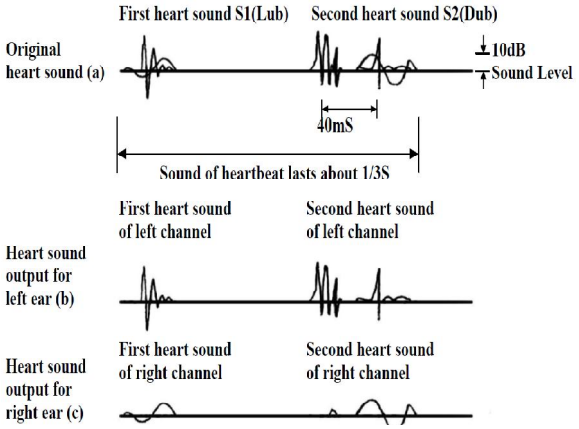


Fig.5: Operation of surroundsound electronic stethoscope

Fig.6 shows the working principal of this system, the pick-up mono input signal from the microphone is applied directly to one end of the left and right channels of the amplifier, while this sound is also applied to an adjustable time delay circuit, then to a phase splitter, which outputs a non-inverted signal to the other end of the right channel amplifier, and a inverted signal to the other end of the left channel amplifier, the input side of the amplifier can be used with a phase adder circuit or a differential amplify circuit.

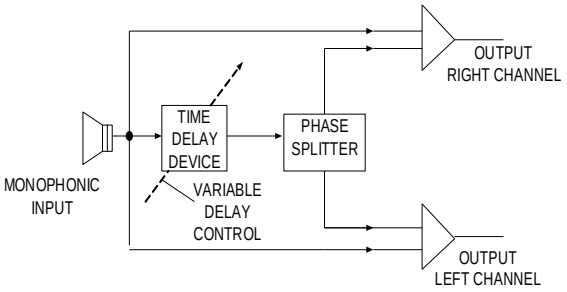


Fig.6: Working principal

This circuit works like follows: in the time delay circuit, the delay controller sets a variable-delay time to generate the delayed audio signal, the right channel is connected to the amplifier through a phase splitter, in which the delayed signal is not heard as a separate sound in the right ear, but changes the apparent sound. At the same time, the phase splitter supplies an inverting delay signal for the left channel amplifier, and therefore, the combined signal with the direct signal from the microphone is different from original and also different from the sound of the right channel. The length of the time delay is always less than 40 ms, the direct sound and the delay sound is perceived as separated sound like the original and its echo.

Fig.7 illustrates a specific circuit, wherein the phase splitter is in the form of a transistor, the transistor is not used as a voltage amplifier in this circuit. A sound signal inputs to the base of this transistor, which is time-delayed signal from time delay device, as adjusted by a variable delay control, then output the inverse and non-inverse signal from the collector and the emitter circuit.

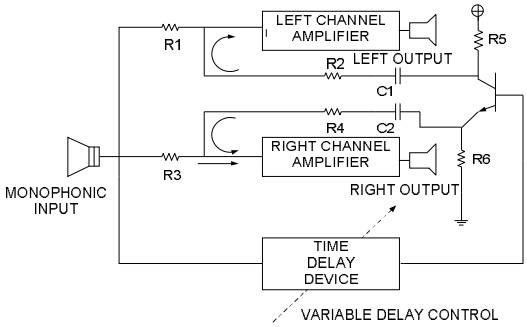


Fig.7: Circuit diagram of one implementation

An inverted signal output from the collector of the transistor passes through a capacitor C1 to the R1 and R2 adder circuit, then to the input of left channel amplifier. While a mono signal obtained from the microphone is also applied through resistor R1 to the input terminal of the same amplifier. Since the signal output from the collector of the transistor has been reversed, the result in the input of the left channel amplifier produces a difference signal. This combined output acoustic signal corresponding to a subtraction or loss of specific frequencies in the left channel.

The non-inverted signal from the emitter passes via the coupling capacitor C2 to the resistive adder circuit of right channel and the microphone sound via a resistor R3 is added simultaneously. The combined output signals represent an amplitude increase at a particular frequency. The net effect on the listener is a perception of separation of the original sound into two different frequencies for each ear, so to gain a depth and spatial quality to the sound.

The frequency of the separating sound depends on the length of the delay time. For example, a time delay of 0.5 ms generates frequency peaks of right-to-left separation of around 1000 cycles per second. The longer time delay produces lower frequency tone separation. Therefore, the frequency at which peak distinction is produced can be selected by adjustment of the variable time delay control.

2.3. System Implementation

Based on the above theories, the architecture of the electronic stethoscope system will be developed as shown in the following diagram, which includes mainly a pre-filter amplifier, a digital sound processing unit, ADC/DAC converters, and some peripheral components (**Fig. 8**, **Fig. 9**).

This electronic stethoscope systemincludes soundsignal capture, sound processing unit and sound display/playback circuitry.The sound capture circuit consists of a microphone, an analog gain amplifier/filter, analog to digital converter circuit; the sound processing unit consists of a microprocessor, digital filtering, digital delay circuit, phase splitter, peripheral components control circuit and other components; the sound display/playback circuit consists of a graphic LCD display, memory card, USB interface, and digital to analog conversion circuit.

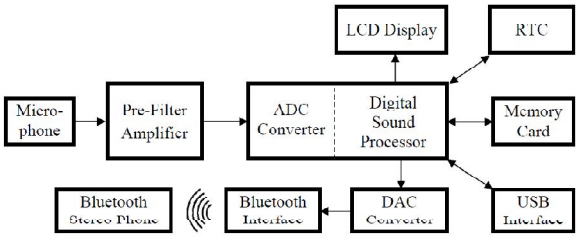


Fig. 8: System components

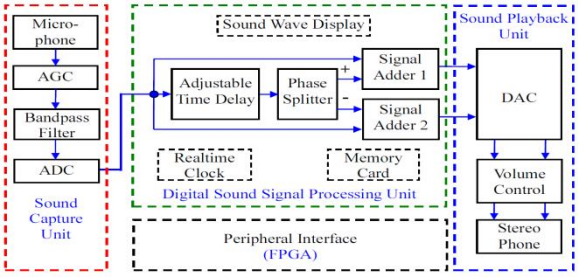


Fig.9:Function blocks of electronic stethoscope

The body organs'weak soundsignalis captured by the microphone, passes throughthe automatically gain control amplifierand thefilter circuit,then connected to the input of a sound A/D converter to the microprocessor for subsequent digital processing, and finally a clear reception of audio signal output from the stereo headphonefor the human ear. This sound signal will also be displayed on a graphic LCD panelas in **Fig.5**, which assists the clinician to determine the cause.

2.3.1 Analog Circuits

The frequency of the heart sound is in the range of about 20Hz ~ 120Hz in general. Therefore, the lower cutoff frequency f_L of the filter is set to 20 Hz, while the higher cutoff frequencyf_H set to 120Hz. **Fig.10** shows a high Q band-pass filter using an operational amplifier.

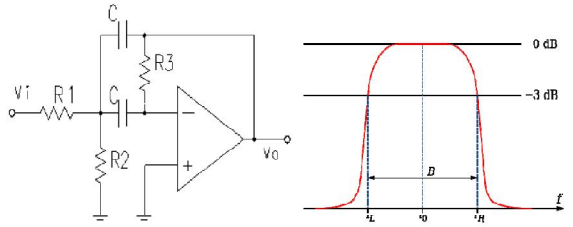


Fig.10: High-Q band-pass filter

The transfer function of this circuit is $H_{center}=\frac{R_2}{2R_1}$. frequency *fo* and the bandwidth *B* of the filter can be calculated as follows:

Radian frequency	Hertz
$\omega_o=\frac{1}{C\sqrt{(R_1\|\|R_2)R_3}}$	$f_o=\frac{\omega_o}{2\pi}$
$\beta=\frac{2}{CR_3}$	$B=\frac{\beta}{2\pi}$

where $R_1\|\|R_2=\frac{R_1R_2}{R_1+R_2}$

The AGC (Automatic Gain Control) automatic gain compensation function is a closed-loop electronic circuit, a negative feedback system, so that the gain of the amplifier circuit is automatically adjusted with the signal intensity. It can adjust the microphone received volume automatically, ensuring the output signal to a certain level to solve the variation problem of the input sound.

The AGC circuit can be divided into an adjustable gain amplifier and a control voltage generating circuit. The adjustable gain amplifier circuit located in forward amplification path, the gain changes with a control voltage; the control voltage detector circuit has a low-pass smoothing filter. The control voltage Vc is generated through a detector by filter out the low frequency noise components from the output signal Vout, when the input signal Vin is increased, the Vc and the Vout are also increased, the increase of Vc causes the decrease of the gain of the amplifier.The amount of the output signal change is less than the significant amount of the input signal change; this achieves the purpose of automatic gain control (**Fig. 11**). **Fig. 12** is a circuit diagram of a specific implementation.**Fig. 13**shows a testing result of this circuit, the signal output level is fixed at 1.26Vp-p, while the input signal varies from 60mVp-p to 20Vp-p.

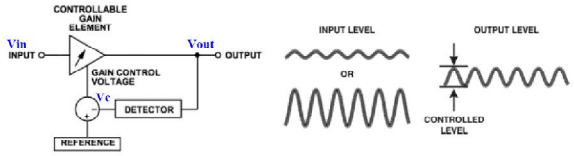


Fig.11: Working principle of an AGC controller

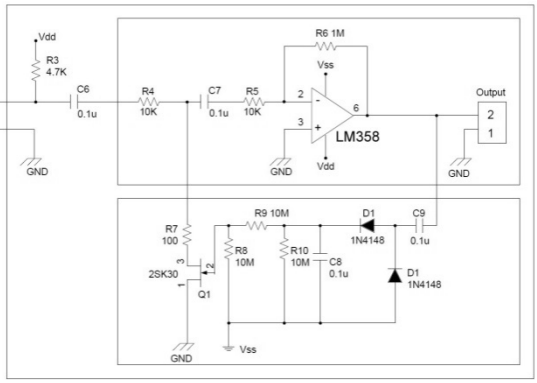


Fig.12:An AGC controller circuit

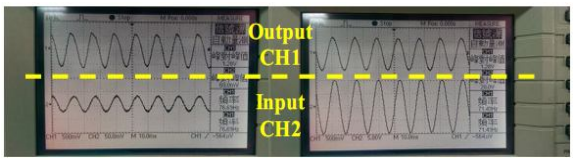


Fig.13: Test result of the AGC circuit

2.3.2 Digital audio delay circuit

The function of sounddelay circuitisto delay the sound signaloutput for some time,it can be implemented with a circular buffer inthe digital circuits(**Fig.14**), whichallows data to beupdated continuously, and replacesthe earliermemorydata. The data is initiallywritten in the firstmemory location, then the new data will be writteninto the nextavailable memoryaddress, once the address

pointerreaches the end ofthe circular buffer, it will point to the firstaddressmemory immediately.Storing datain thiswayis often settothe available buffer2^{n}samples, and theaddress pointercan bedesigned with a simpleN-bit counter.

Thework of a delay circuit is shown in **Fig. 15**with a write address Xn and a read address pointer Yn. Xn pointer indicates the current write address of the buffer, Yn pointesthe data to be read out after a time delay, the relationship can be expressed as *Y[n]=X[n-t]*, where *t* corresponds the time difference between these two samples.

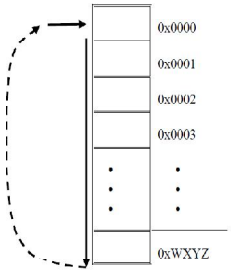


Fig. 14: Structure of ofcircularbuffer

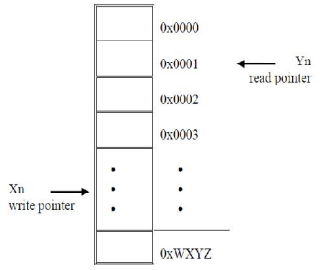


Fig. 15: Working method the delaycircuit

Due to the sampling frequency of the digital audio stream is the same for data written and read, so the *t*value isthe delay time multiplied by the reciprocal of the sampling frequency. It should be noted here that *t > 1* and*t* <N, N is the total number of samples stored in the circular buffer.

2.3.3 Digital Filter

The filter circuit is a way of passing specific frequency, while the other frequency signals are suppressed. Digital filtersare constructed mainly with the delay lines, adders and multipliers.

According to the time-domain impulse response function digital filters can be divided into Infinite Impulse Response (IIR) and Finite Impulse Response (FIR) filter.

The Finite Impulse Response filter is composed of finite length inputs, and its non-recursive design is very stable. To meet the requirements of amplitude-frequency response a strictly linear phase characteristics can also be obtained at the same time, and therefore, it is widely used in the field of high-fidelity signal processing. Its transfer function can be written as:

$$H(z) = \sum_{k=0}^{N-1} h(n) z^{-k}$$

The output equation of FIR filter is described as:

$$y(n) = \sum_{i=0}^{L-1} x(n-i)h(i) \qquad (1)$$

where: x(n) is the input sample sequence; h(n) is the filter coefficient; L is the filter order; y(n) represents the filter output sequence and is the sum of products of x(n) andh(n). The basic structure of the FIR filter shown in **Fig. 16**:

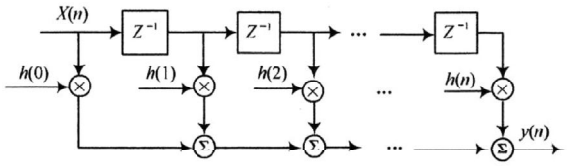


Fig. 16: FIR digital filter circuit

The output of a fourth-order filter system can be expressed as:

$$y(n) = h(0)x(n) + h(1)x(n-1) + h(2)x(n-2) + h(3)x(n-3) \qquad (2)$$

To realize this filter circuit 3 delay lines, 4 multipliers and 1 four-input adder are required.

III. RESULTS AND DISCUSSION

Fig. 17 shows thesurround sound electronic stethoscopeusing an Altera DE2i-150 FPGA development board.A embedded platform that combines an Intel embedded N2600 processor with an Altera Cyclone IV GX FPGA, DE2i-150 is equipped with around 150K logic elements with ultimate flexibility in terms of reconfiguration ofactual hardware circuitry and intellectual property as well as on-board multimedia peripherals.

The Intel Atom processor and the FPGA device are linked together via two high-speed PCIe interfaces, which offer a powerful hardware-software co-development environment.

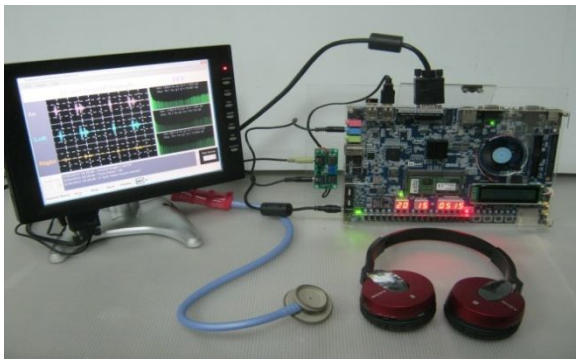


Fig.17: Realization of surround sound electronic stethoscope

Fig. 18 shows the software developingenvironment using Microsoft Visual Studio, which enables a quick development of the graphical human-machine control interface, and provides rich libraries for display signal in time and frequency domain.

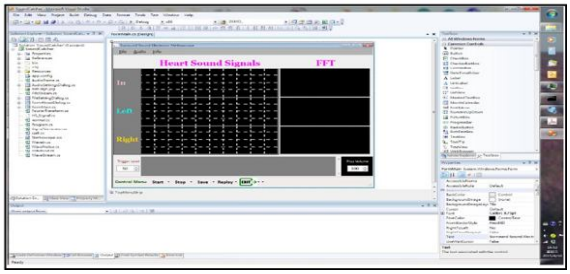


Fig.18: Develop environment of a graphic user interface

Fig. 19 showsan example of a single heartbeat. This device converts the incoming signal to left and right output channel by setting a time delay of 100 samples (about 2.3mS). A FFT spectrum display of each channel is also provided for signal analysis in frequency domain.

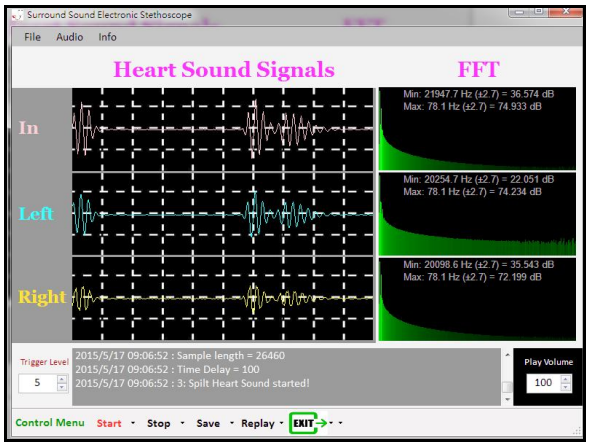


Fig.19: Test sample of a signal heart sound

CONCLUSIONS

The surrounding sound produced by this new electronic stethoscope will provide higheraccuracyfor doctors to listen to the heartbeatwith the following advantages:

1. The analog audio pre-processing unit can reduce or eliminate unnecessary noise, and enhancethe sampling sound quality,

2. The digital sound processing unit can distinguish the heart sound that is difficult for the human ear to hear.The sound waveform can also be displayed, and available for discussions with many viewers,enabling physicians to diagnose them more accurately.

3. The capturedsound signal can be stored to track the development of the physical symptoms.

4. Use the electronic stethoscope to record and playback of the heart sounds is benefit for physicians during examination or for teaching of cardiovascular physiology during lecture.

5. In addition to medical diagnosis, it can also be used for industrial applications, for example, to detect afine engine noise or machinery abnormal vibration and so on.

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