Master's Thesis
석사 학위논문

Signal Processing Design for Cochlear Implant System Using Piezoelectric Sensor

Jae Wook Cho (조 재 욱, 曺 宰 頊)

Department of Information and Communication Engineering

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Advisor : Professor Ji-Woong Choi

Co-advisor : Professor Hui Sup Cho

by

Jae Wook Cho
Department of Information and Communication Engineering
DGIST

A thesis submitted to the faculty of DGIST in partial fulfillment of the requirements for the degree of Master of Science in the Department of information and communication engineering. The study was conducted in accordance with Code of Research Ethics[1]

11. 15. 2013

Approved by

Professor Ji-Woong Choi (Signature)
(Advisor)

Professor Hui Sup Cho (Signature)
(Co-Advisor)

[1] Declaration of Ethical Conduct in Research: I, as a graduate student of DGIST, hereby declare that I have not committed any acts that may damage the credibility of my research. These include, but are not limited to: falsification, thesis written by someone else, distortion of research findings or plagiarism. I affirm that my thesis contains honest conclusions based on my own careful research under the guidance of my thesis advisor.

Signal Processing Design for Cochlear Implant System Using Piezoelectric Sensor

Jae Wook Cho

Accepted in partial fulfillment of the requirements for the degree of Master of Science.

12. 03. 2013

Head of Committee ___________________ (Signature)
Prof. Ji-Woong Choi

Committee Member ___________________ (Signature)
Prof. Jae-Eun Jang

Committee Member ___________________ (Signature)
Prof. Hui Sup Cho

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ABSTRACT

Today, the cochlear implant system is the most successful auditory prosthesis for the patients suffering hair cell loss or damage. The current cochlear implant system consists of an implantable electrode array for the stimulation and an external signal processor including a microphone and a battery. However, the conventional cochlear implant system is composed of a heavy signal processor. Furthermore, the exposure of an external signal processor causes a cosmetic problem to the patients. Therefore, the cochlear implant system is required to reduce size and power consumption.

In the thesis, to realize the implementation of an entirely implantable cochlear system, the signal processing system is studied when a piezoelectric sensor is employed. When a mechanical stress is applied, the piezoelectric sensor produces an electrical charge. The acoustic sounds picked up by the piezoelectric sensor are amplified and transmitted to the microprocessor. The microprocessor used the CC430F6137. This system is a multi-channel system and applied the continuous interleaved sampling (CIS) algorithm to make a charge-balanced biphasic pulse. Also, this system is referred to as miniaturized system comprised of sensing, processing, and actuating functions which are combined onto a single board. Building a complete printed circuit board (PCB) involves hardware (H/W) and software (S/W) design. The proposed signal processing system embedded with PCB has been tested and verified for future animal experiment.

Keywords: cochlear implant, piezoelectric sensor, signal processing, small size, low power consumption

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Ⅰ. INTRODUCTION

In the world, there are a lot of people who have a sensorineural hearing loss. The sensorineural hearing loss is a type of deafness which is caused by the damage on hair cells of cochlea in inner ear. The hair cells convert acoustic sounds to electric signals that stimulate auditory nerves. As one of the treatments for the sensorineural hearing loss, a cochlear implant is recently used [1]. The current cochlear implant system consists of an implantable electrode array for the stimulation and an external device including a microphone, a signal processor and a battery. The acoustic sound is detected and is processed with respect to the frequency by the external device. The processed charge-balanced electrical signals are transferred through a wireless link. Then, the auditory nerves are stimulated through the electrode arrays inserted in the cochlea [1]. Even though current cochlear implant system is an optimal solution to restore the hearing loss, it has two technical shortcomings up to now.

The size and power consumption are the main disadvantages of current cochlear implant systems. Above all, patients want small devices that can be worn on the head due to cosmetic problem. In addition, they do not want to recharge the batteries often. For these reasons, an entirely implantable cochlear system is proposed by artificial cochlea. The functions of artificial cochlea are not only the conversion of acoustic wave to electric signals but also the frequency selectivity [2, 3]. In the cochlea of inner ear, the basilar membrane has an important role for the frequency selectivity. The resonance frequency of membrane change along the place because of varying mechanical boundary conditions. If the resonance frequency at a local place match to certain frequency of acoustic wave, the place vibrates with relatively large amplitude based on the resonance theory. Then, the vibration stimulates hair cells at the resonated place. As a result, the frequency of acoustic wave is recognized as the difference in tones. To realize the frequency selectivity, micro electro mechanical system (MEMS) devices have been reported. Hongsoo Choi et. al. [4] developed a piezoelectric acoustic sensors with the function of frequency selectivity by the use of resonance of cantilever arrays. Those sensors were evaluated in the atmospheric environment. Eun Sok Kim et. al. [5] developed a resonance-enhanced piezoelectric microphone array for prefiltered acoustic sensing. The piezoelectric acoustic sensor realizes both the frequency selectivity and the conversion of acoustic sound to the electric signal without an external energy supply. This piezoelectric sensor is called an artificial basilar membrane (ABM).

In this thesis, the manufactured printed circuit board (PCB) is designed as a prototype model to test the basic concept of artificial basilar membrane (ABM) for the development of the entirely implantable cochlear system. The board mechanically decomposes the frequency of acoustic sounds instead of electric band-pass filters for smaller size and low power consumption.

The organization of this thesis is as follows. Section 2 gives an introduction to the physiology of hearing and the brief overview of history. Section 3 introduces the principles of the typical cochlear implant system and the entirely implantable cochlear system, and describes the biphasic signal. The piezoelectric sensor is essentially used in the proposed entirely implantable system. Section 4 describes the system design and development in an aspect of the circuit design and shows the results of manufactured PCB, debugging and test. At last, the conclusions will be given in Section 5.

Ⅱ. BACKGROUNDS

2.1 Auditory System

The human auditory system collects acoustic sounds and converts it to electric signals. The sounds from 20Hz to 20kHz can be detected. As shown in Figure 1, human ear can be divided to three regions referred to as outer ear, middle ear, and inner ear [6].

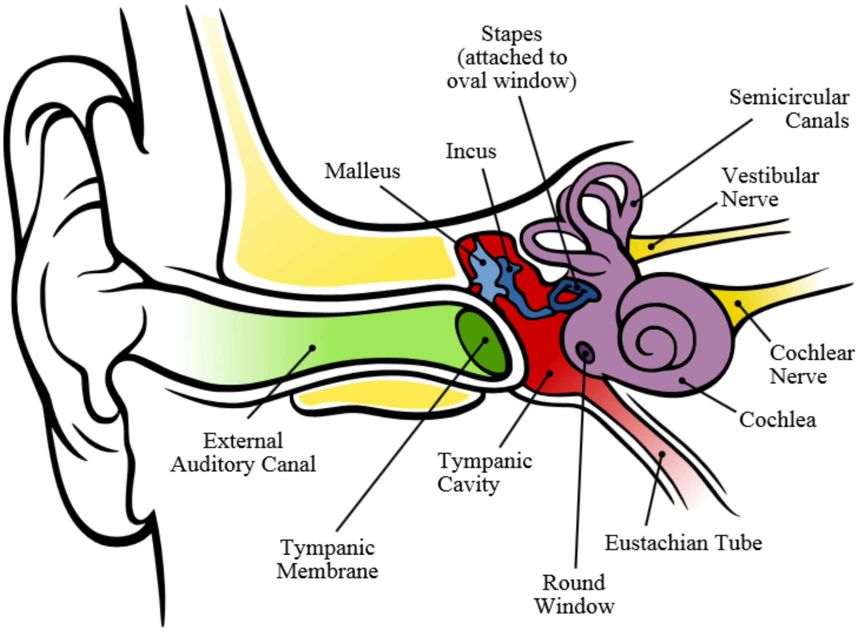

Figure 1. Anatomy of the human ear [6]. The outer ear includes the pinna, the ear canal, and the ear drum. The middle ear includes the three bones. The bones are the malleus, the incus, and the stapes. The inner ear includes the cochlea, vestibule, and three semicircular canals.

The primary purpose of the outer ear collects acoustic sounds and transfers it to the cochlea, much like a radio antenna. The acoustic sounds have been conducted into the auditory channel and vibrates the tympanic membrane. The three small bones of the middle ear amplifies the acoustic sounds, much like a lever system. The cochlea is a small spiral-shaped bone inside the inner ear that separates the incoming acoustic sounds based on their resonance frequencies. The cochlea is filled with a thick fluid and is separated to two chambers by the basilar membrane. Incident acoustic sounds collected by the middle and outer ear cause the basilar membrane to vibrate, conducting a travelling wave down the length of the cochlea. As shown in Figure 2, different regions of the basilar membrane resonate at different resonance frequencies because of the variations in the characteristics of membrane along its length. As a result, high frequency waves die out closer to the entrance of cochlea and lower frequency waves travel further down the basilar membrane.

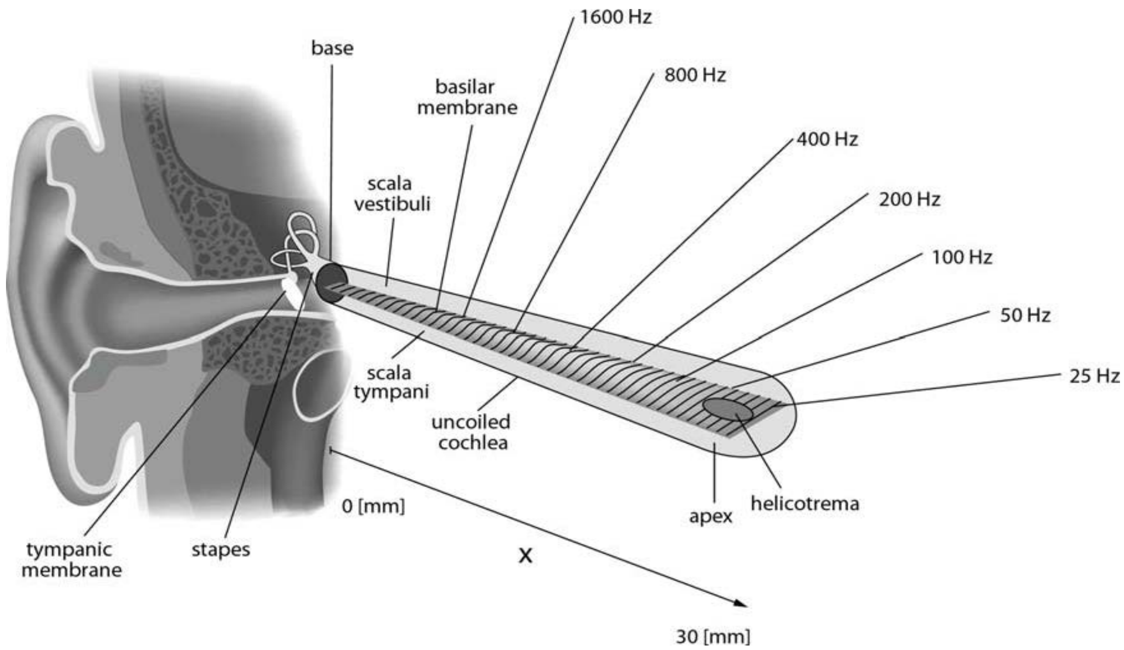

Figure 2. Uncoiled cochlea with basilar membrane [7]. The position x of the maximal amplitude of the travelling wave corresponds in a 1-to-1 way to a stimulus frequency.

A traveling wave resonates on the membrane before dying out, causing the auditory nerve to be stimulated in a specific location for its frequency [7]. The cochlea is a filter-bank that separates incoming acoustic sounds due to their resonance frequency. The frequency resolution of the cochlea is very fine, limited only by the number of individual regions where the auditory nerve can be stimulated by resonant traveling waves. Each stimulation sites correspond to the group of hair cells. The cochlea has separated the frequency components along its basilar membrane and the hair cells transfer the information of acoustic sounds to the brain. As shown in Figure 3, when the basilar membrane is excited by a traveling wave, hair cells attached to the membrane are excited by a displacement force.

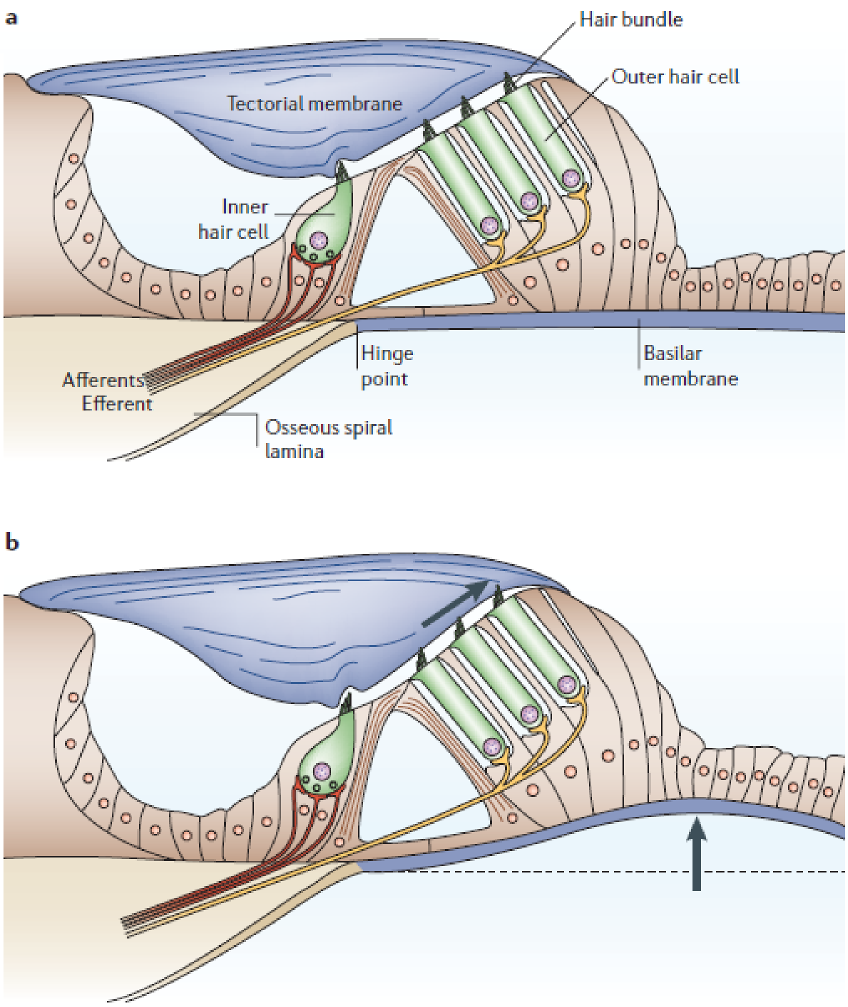

Figure 3. Cellular structure of the sound detecting organ of Corti [8]. (a) Transverse section through a middle trun of the cochlea, showing the organ of Corti, an assembly of an assembly of intricately shaped supporting cells and inner and outer hair cells supported by a flexible basilar membrane. (b) Upward displacement of the basilar membrane stimulates the hair cells by bending their stereociliary bundles against the acellular tectorial membrane. Because of the point about which the basilar membrane hinges, the inner hair cells must be stimulated mainly by motion of the tectorial membrane. Outer hair cells have both sensory and motor capabilities and possess electromotility that underlies the cochlear amplifier. They have a sparse afferent innervation and are contacted mainly by efferent nerves, which regulate the electromotility and influence cochlear sensitivity.

This displacement force opens mechanically-gated ion channels in the hair cells, which depolarizes nearby neural cells and induces a neural oscillation [9]. This neural impulses is known as an action potential, and propagates the auditory nerve into the brain through the auditory processing. The action potential is a response, and always has the same form like a digital bit with the peak amplitudes of acoustic stimulation [10]. The process of normal hearing begins with the occurrence of the acoustic sounds. When this acoustic sounds stimulate the ear, the sound is heard. In the human ear, the sounds are transmitted through each process along the auditory system.

2.2 Hearing Loss

As many as 0.3% of the population in the United States are born deaf or exhibit a hearing loss. As an older people, the percentage of a hearing loss increases. It is estimated that 17% of adults in the United States exhibit a hearing loss. 30% of people over the age of 65 years exhibit a hearing loss, and this percentage increases to 47% of people 75 years of age and older [11]. If the hair cells are damaged, the auditory system has no way of transforming sound to neural impulses, resulting in hearing loss. The hair cells can be damaged by certain diseases or many other cases. Figure 4 shows pictures of normal and damaged hair cells.

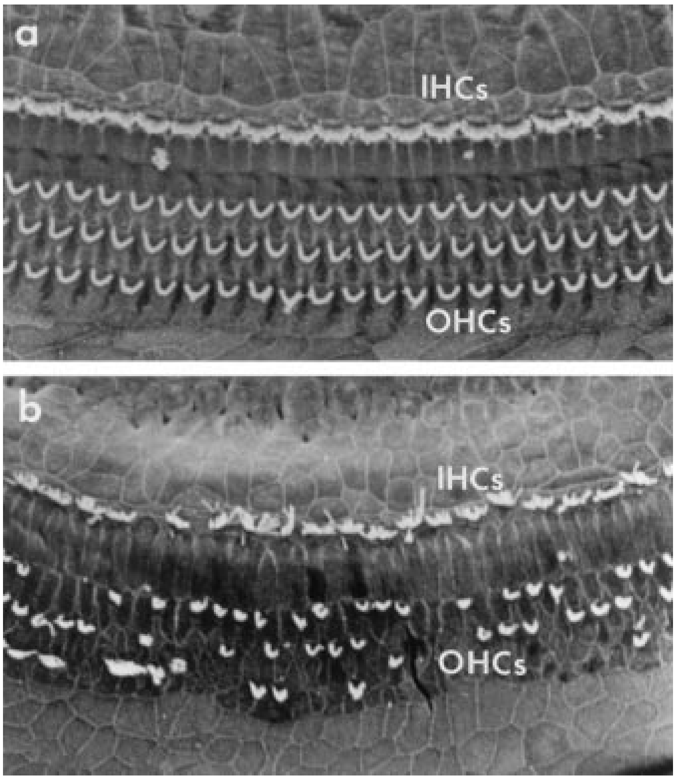

Figure 4. (a) Normal (b) Damaged cochlear sensory epithelium [12]. In the normal cochlea, the stereocilia of a single row of inner hair cell and three rows of outer hair cells are present in an orderly array. In the damaged cochlea, hair cells are missing, and stereocilia are abnormal, leading to hearing loss.

Damaged hair cells can lead to degeneration of adjacent auditory nerves. If a large number of hair cells or auditory nerves are damaged, then the condition is called profound deafness. Some research has shown that most common cause of deafness is the loss of hair cells, rather than the loss of auditory nerves [12]. This way very encouraging for cochlear implants because the remaining nerves could be excited directly through electrical stimulation. A cochlear implant is based on the idea of bypassing the normal hearing mechanism and electrically stimulating the remaining auditory nerves directly. Therefore, the sensorineural hearing loss can be viewed as a biological disorder caused by problems of hair cells within the cochlea. The typical treatments for sensorineural hearing loss are based on electronic technologies such as electrical stimulation of the surviving spiral ganglion neurons using cochlear implants. The cochlear implant has not only provided useful hearing to more than 120,000 deaf people, but also is the most commercially successful auditory prostheses to restore deafness [1]. The success of cochlear implant can be attributed to the combined efforts of scientists from various disciplines including bioengineering, physiology, and signal processing. The signal processing, in particular, plays an important role by converting acoustic vibrations into electrical stimuli signals to be transmitted to the auditory nerve where the signal is separated into multiple frequency bands.

2.3 History of Cochlear Implant Systems

Around 1790, it was discovered that electrical stimulation of the auditory system can create a perception of sound. The discovery was made when Alessandro Volta placed metal rods in his own ears and connected them to a 50V circuit, hearing a noise [13]. Other experiments were performed until electrical, sound amplifying hearing aids were developed in the 20th century. Few people realize that cochlear implantation also has a long history, going back more than half a century. The first direct stimulation of an acoustic nerve with an electrode was performed in Paris in 1957 by Andrr Djourno and Charles Eyries [14]. Having placed wires on nerves exposed during an operation, they reported that the patient heard sounds like “a roulette wheel” and “a cricket” when a current was applied. After implanting two patients, a dispute over the commercial development of their discovery put an abrupt end to their collaboration [15]. In the 1970s, the center of activity shifted from Paris to Los Angeles. There, in 1972, William House and Jack Urban developed the first cochlear implant, which was registered by the American Food & Drug Administration in 1979 as a clinical appliance [16]. This single-channel implant permitted adequate stimulation of the cochlear nerve [17, 18]. The results were encouraging; by allowing patients to pick up ambient sounds, the implant served as an aid to lip-reading. During the 1970s, Graeme Clark took the development of cochlear implants in a new direction by stimulating the cochlea at multiple points [19]. The advantage of multi-channel cochlear implants, which became more widely available in the 1980s, was far better speech discrimination. These implants follow the tonotopic distribution of the cochlea and the auditory nerve as described by Von Helmholtz in 1863 and Von Bekesy in 1960 [20, 21]. The latter discovered the tonotopic tuning that exists along the length of the organ of Corti. Essentially, high-frequency sounds cause motion in the organ of Corti at the base of the cochlea, whereas low-frequency sounds cause motion at the apex of the cochlea, several centimeters down the length of the organ of Corti. Throughout the 1990s, the external components, which had to be strapped to the body, became smaller with advances in miniaturization electronics. Today most adults and school-age children wear a little behind-the-ear (BTE) speech processor about the size of a power hearing aid.

Ⅲ. COCHLEAR IMPLANT SYSTEM

The cochlear implant system is an auditory prosthesis that has been used as a normal treatment for the patients obtained sensorineural hearing loss. The cochlear implant system must perform three basic roles. First, it must detect a wide range of the acoustic sounds. Second, it has to process the acoustic sounds into the charge-balanced electrical signal forms for each electrodes. Third, the implanted part must stimulate the auditory nerves of the cochlea with the pulses of current. Figure 5 shows how the implanted system is configured in the human ear on the body.

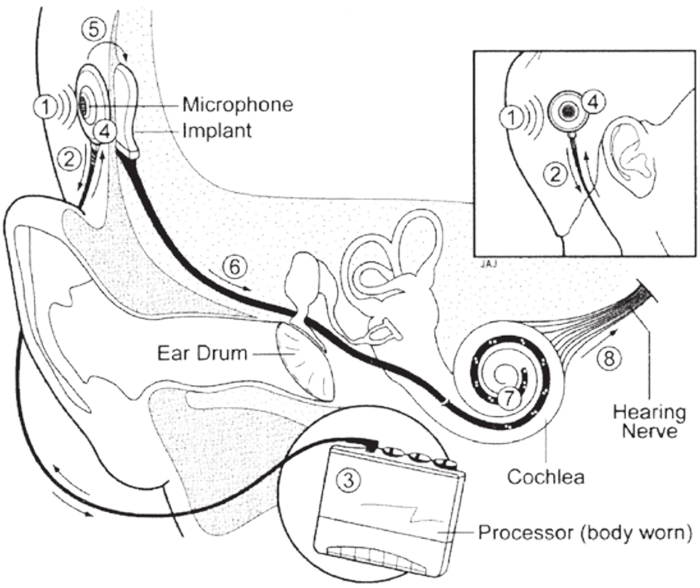

Figure 5. Components in a typical cochlear implant system [22]. A microphone picks up the sound and sends it to the speech processor that is worn behind the ear. The speech processor converts the sound into a digital signal. The signal travels back to a headpiece that contains a coil transmitting coded radio frequencies across the skin. The headpiece is held in place by a magnet attracted to the implant on the other side of the skin. The implant contains another coil that receives the radio frequency signal and also hermetically sealed electronic circuits. The circuits decode the signals, convert them into electric currents, and send them along wires threaded into the cochlea. The electrodes at the end of the wire stimulate the auditory nerve connected to the central nervous system, where the electrical impulses are interpreted as sound.

3.1 Typical Cochlear Implant System

A typical cochlear implant system essentially consists of two parts that include an external part and an internal part. External part consist of three parts: Microphone which picks up the acoustic sounds from the environment; Signal processor, which selectively filters the sounds to select audible speech, and sends the charge-balanced electrical signals to the transmitter through a cable; Transmitter, which is a magnetic coil placed behind the ear, and transmits the processed signals to the internal device by electromagnetic induction. Internal part consist of two parts: Receiver and stimulator, which sends the signals to the electrode arrays through an internal cable; Electrode arrays, which send the charge-balanced electrical signals to the auditory nerves in the scala tympani and then directly to the brain through the auditory nerve system [1].

3.1.1 Microphone & Signal Processor

Human can usually hear the sounds in the range of frequency from 20Hz to 20kHz, but most cochlear implant systems provide the sounds of the frequencies between 300Hz and 10kHz due to electrode limitations and other factors. A function of the microphone is to detect the acoustic sounds. A requirement of the good microphone is to provide the sounds of wide input range to the signal processor. A function of the signal processor is to convert the input signals to certain patterns of electrical stimulation. Ideally, the outputs of the signal processor must be represented as an information that can be perceived by patients. A current processor is powered with batteries. The battery life typically exceeds 12 to 16 hours, allowing patients to use their devices during the waking hours without the need for recharging or replacing the batteries [1]. Adequate battery life is made possible through the use of integrated circuit technology, particularly digital signal processing (DSP) chips that have become available within the past 10 years. Advances in battery, integrated circuit, and DSP chip technologies have been driven by commercial markets. The current cochlear implant system require the better batteries and power efficient chips. This current processor is wearable form which is more cosmetic and more convenient until today. The cochlear implants have made possible progressively smaller and more capable signal processors. Even greater reductions in size and increases in capabilities may be available in the near future. Entirely implantable systems may be available within the several years.

3.1.2 Transmission Link

The purpose of a transmission link is to transfer the speech information from the external signal processor to the implanted stimulator through wireless technology. An advantage of this link is that the skin is closed over the implanted components, which may reduce the risk of infection. A disadvantage is that a limited amount of information can be transmitted across the skin with a link. A typical cochlear implant system is a telemetry system, which is included to allow electrode voltage waveforms to be monitored externally after the internal device is implanted. The typical cochlear implant systems use an inductive link [23]. Power and data are transmitted by the inductive link. The internal device also incorporates a back-telemetry mode for measurement and feedback. The central part of the internal device is an application specific integrated circuit (ASIC) chip with mixed analog and digital components. The circuits of the chip are designed to operate from a variable supply voltage within a range of 2.5V to 5.5V. In practical application, the minimum necessary supply voltage is of interest, since it determines the power consumption of the device.

3.1.3 Electrode Array

The electrodes are implanted inside the cochlea. They must be bio-compatible and mechanically stable in order to sustain vibrations from head movement [1]. Their purpose is to stimulate the cochlea with electrical current. Electrode stimulation can happen with either an analog or pulsatile signal, depending on the type of signal processing algorithm. Pulsatile stimulation can happen in one of several ways: monopolar, bipolar, or dual electrode configurations. Monopolar, or common ground, stimulation fires one electrode at a time with reference to a common ground electrode for any stimulating site. In bipolar stimulation, each firing electrode has a nearby ground electrode to provide a shorter path for current and a more localized stimulation. For dual electrode stimulation, two electrodes in close proximity are fired simultaneously in order to stimulate nerves around both electrodes. The choice of stimulation methodology depends on the signal processing algorithm. Depending on the location of the return or ground electrode, electrode configurations can vary in the mode of monopolar and bipolar. In the monopolar configuration, each electrode is stimulated with reference to a remote electrode. In the bipolar configuration, one electrode is stimulated with reference to another nearby electrode. Different pairs of electrodes are used to stimulate different sites along the electrode array.

3.2 Biphasic Pulse

Information is presented either in analog or pulsatile form. In analog stimulation, an electrical analog of the acoustic waveform is presented to the electrode. In multichannel implants, the acoustic waveform is band-pass filtered, and the filtered waveform are presented to all electrodes simultaneously. One disadvantage of analog stimulation is that its simultaneous action may cause channel interactions. In pulsatile stimulation, the information is delivered to the electrodes using a set of narrow pulses are extracted from the envelopes of the filtered waveforms. The advantage of this type of stimulation is that the pulses can be delivered in a non-overlapping mode, thereby minimizing channel interactions. Pulse rate has been found to affect recognition performance. High pulse rates tend to yield better performance than low pulse rates. As shown in Figure 6, when current is applied to an electrode array, the waveform flows from the negative to the positive pole. Typical stimulation current is 10uA up to 1.6mA to avoid remaining charge in the tissue. Typical stimulation rate of biphasic pulse is 1,000pps [24].

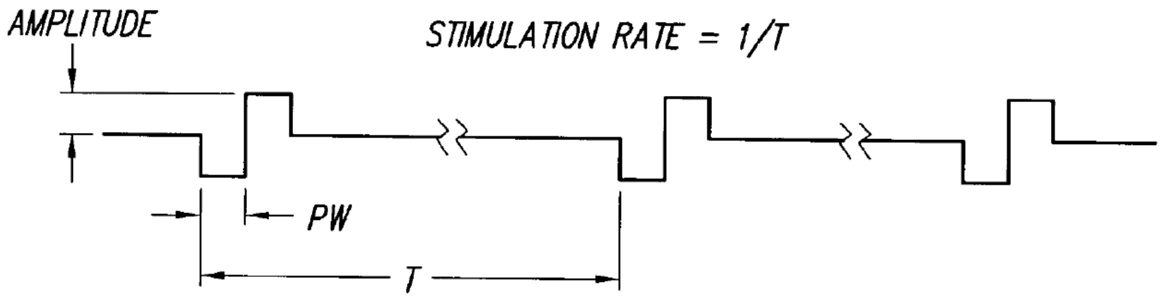

Figure 6. Schematic of biphasic pulse [25]. It is a current stimulation waveform that defines the period (T), the stimulation rate (1/T), the biphasic pulse width (PW), and the biphasic pulse amplitude associated with electrical stimuli.

3.2.1 Continuous Interleaved Sampling Algorithm

In the past, the stimulator transmitted the analog waveform to each electrode with frequency depending on their location in the cochlea. The input signal was filtered using analog band-pass filters to target the correct part of the cochlea with the correct frequency band. However, this scheme created an electric field of varying magnitude around each electrode. As a consequence, the interference was present due to electrode signal coupling among each other. To solve this problem, a continuous interleaved sampling (CIS) is emerged [26]. The CIS developed by Dr. Blake Wilson at the Research Triangle Institute (RTI), is the most common algorithm used by cochlear implant system signal processing today. The CIS has the best speech comprehensive performance of any other commercially algorithm with existing electrode arrays [26]. Figure 7 is a block diagram for an 8 channels CIS processor.

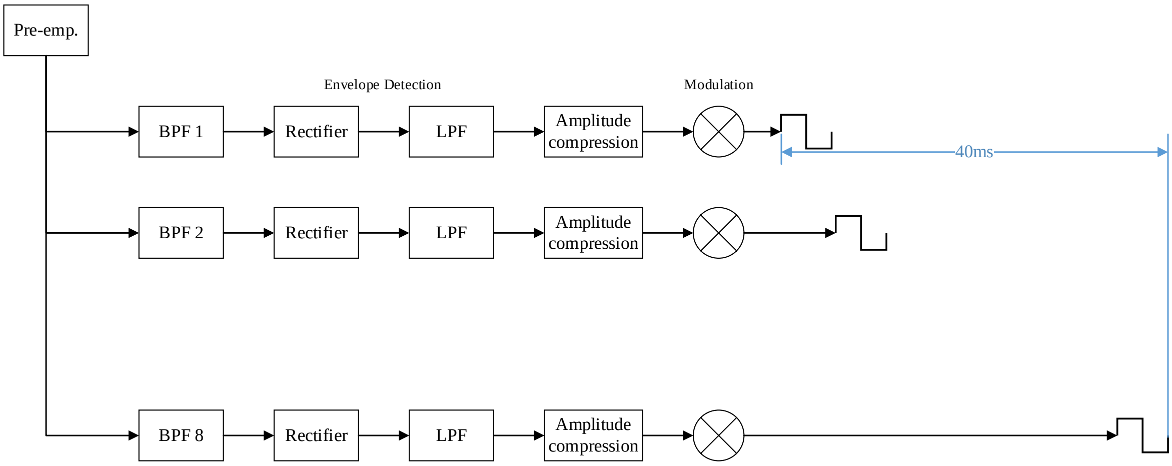

Figure 7. Block diagram of the CIS algorithm. The signal is first pre-amphasized and filtered into eight frequency bands. The envelopes of the filtered waveforms are extracted by full-wave rectification and low-pass filtering. The envelope outputs are compressed to fit the dynamic range and modulated with biphasic pulses. The biphasic pulses are transmitted to the electrodes in an interleaved mode.

The input signal is divided into logarithmically spaced bands using band-pass filters. An envelope detector obtains the present energy using a rectifier and a low-pass filter. After the signals are compressed by a nonlinear coefficient map to an audible range, the modulation blocks prevent more than one electrode to be active at the same time [26]. The modulation block ensures that the electrode input signals are modulated with pulsatile periodic phase shifted waveforms, one by one, with amplitude depending on the magnitude of the energy coefficient. As a result, no two electrodes are powered at a given time. The output of each channel is modulated with non-overlapping biphasic pulses. The pulse amplitude, both positive and negative phases, is determined by the amplitude of the signal in the channel. The pulse rate, pulse width, and channel to electrode assignment are all factors that are patient specific and are programmed. These pulses are delivered at a high rate of typically 1,000pps per electrode [24]. It is important to ensure that the pulse rate is higher than twice the cutoff frequency of the LPF envelope detector in order to prevent aliasing effects. Also, the positive and negative of the biphasic current pulse must be balanced because any charge imbalance can cause damage to the inner ear tissue.

3.3 Entirely Implantable Cochlear System

Current technologies of cochlear implants have limited because of its expense, inconvenience, and frequent recharging caused by heavy power consumption. In regard to this point, piezoelectric materials have the feature of being able to generate an electrical current. Due to self-powering property, piezoelectric materials are suited as a replacement for the entirely implantable cochlear system. Several research groups carried out various studies on the artificial basilar membrane mimicking structure [4, 5]. The entirely implantable cochlear implant system is referred to devices comprised of sensing, processing, and actuating functions where two or more of the following technologies are combined. The motivations are the need to increase performance while reducing system size, cost, integration complexity, and power consumption. Building a complete entirely implantable cochlear implant system involves several challenges, as these designs unite not only analog and digital circuit technologies, but also the mechanical, biological, and electrical technologies. A successful entirely implantable cochlear implant system requires integration of piezoelectric sensor and circuit design at the device level.

3.3.1 Piezoelectric Sensor

The entirely implantable cochlear system offers the possibility of greatly simplifying the process of artificial electrical stimulation of auditory nerve fibers within the cochlea. As shown in Figure 8, the entirely implantable cochlear system uses the biomimetic piezoelectric acoustic sensor developed by prof. Hongsoo Choi’s group [4]. This biomimetic mechanical sensor, which is one of principle cores of our proposed cochlear implant system, substitutes the role of microphone and digital signal processor for frequency decomposition of the conventional cochlear implant. Thus, this sensor is helpful to construct a total implantable cochlear system because of its small size and low power consumption.

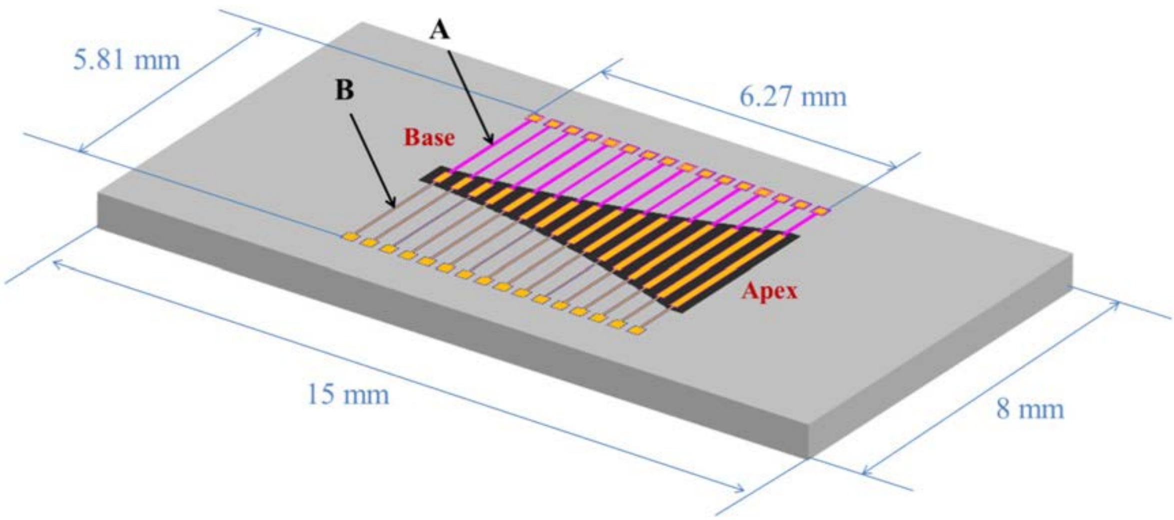

Figure 8. Schematic of an AIN artificial basilar membrane (ABM) [4]. (A) bottom electrode (Mo/Ti). (B) Top electrodes (Au/Ti). The active piezoelectric layer (AIN) was positioned between the two electrodes.

The sensor has a piezoelectric material which can convert mechanical stresses to electrical signals. A piezoelectric sensor developed hear is shown in Figure 9. The sensor has 16 channels and is bonded on the printed circuit board (PCB) with a wire-bonding. The output signal is transmitted through the BNC connectors. It can be easily vibrated by the acoustic sounds. The local resonant frequency of the sensor gradually changes due to the varying mechanical boundary conditions along the length. The local resonant frequency is expected to be decreased as the length increases. Applying the acoustic sounds with a certain frequency to sensor, the local place vibrates with relatively large amplitude due to the resonance. The results mean the frequency selectivity realized by the piezoelectric sensor.

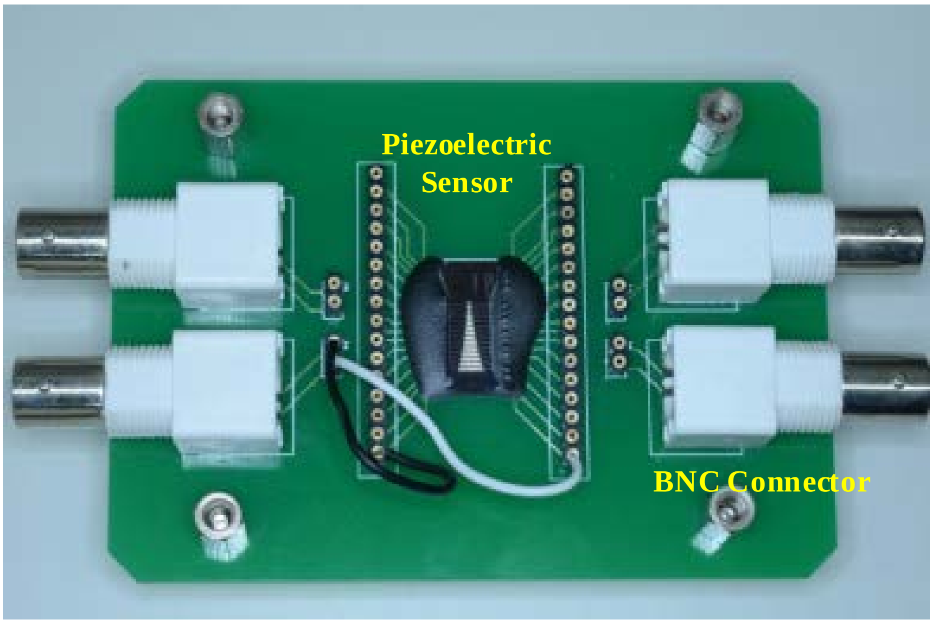

Figure 9. Piezoelectric sensor on board [4]. The piezoelectric sensor is a device that uses the piezoelectric effect to measure pressure by converting it an electrical signal. This sensor has proven to be versatile for the measurement of various processes. It is used for entirely implantable cochlear system.

3.3.2 Proposed System

To realize the implementation of an entirely implantable cochlear system, this system used the piezoelectric sensor in [4]. This system demands a low power consumption and a compact design to enhance the quality of patients' life. To meet the goal, the work proposes this architecture that can be implemented on a single board resulting in small size and low power consumption. As shown in Figure 10, the work focus on low power consumption with increased integration of electrical components into a single board thereby reducing the size. Acoustic sounds picked up by the piezoelectric sensor are carried to the amplifier. The sensor output has to amplify an amplitude of the signal to send to the microprocessor. In the microprocessor, signal processing is performed based on the modified CIS algorithm.

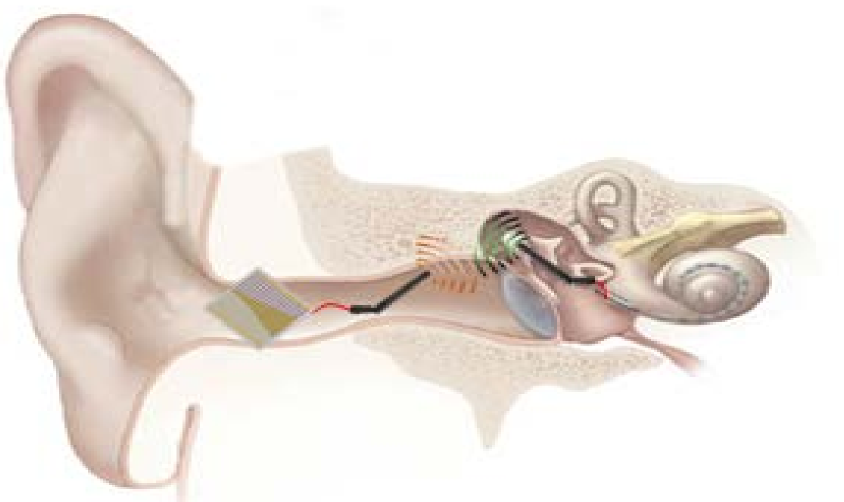

Figure 10. Proposed cochlear implant system. The system consists of two main parts. The first part is an signal processor which can be coupled with the second part, while the second part, an electrode, is an electrode placed into the cochlea to stimulate the auditory nerves. The system incorporates a piezoelectric sensor acting both as an energy harvester and a frequency detector replacing the microphone used in the conventional cochlear implants. The battery is eliminated due to energy harvesting property of the proposed system in the future, while it is currently operated by battery.

Ⅳ. SYSTEM IMPLEMENTATION

The conventional cochlear implant system is essentially composed of a microphone and a signal processor. Acoustic sounds are detected and processed by these components. As the proposed entirely implantable cochlear system, the microphone is replaced by the piezoelectric sensor in [4]. A primary feature of the piezoelectric sensor is to perform the band-pass filtering at different band ranges of each channel. Thus, a signal processor is required to process the rest of the band-pass filtering.

In the thesis, the main difference is the implementation of the signal processing system with the piezoelectric ABM sensor in [4]. The system is designed in a single board. It is rather designed for the laboratory use. For now, it should be possible to use it as an experimental board. Despite the experimental use, the board should be easy to use. To provide the flexible use, the board should be configurable in the software. For the testing purpose, all the hardware parts should be accessible.

4.1 Block Diagram

Figure 11 describes the block diagram of system. The system works as a signal processor which translates the acoustic sounds into the biphasic pulses. The biphasic pulse has an amplitude and a width of the pulse from the acoustic sounds. At the signal processor, the acoustic sounds are quantized by the analog to digital converter (ADC) with a fixed sampling frequency. At the digital to analog converter (DAC), the integrated signals are reconstructed as the biphasic pulses. And the biphasic pulses are transmitted to the actuator and the electrode array.

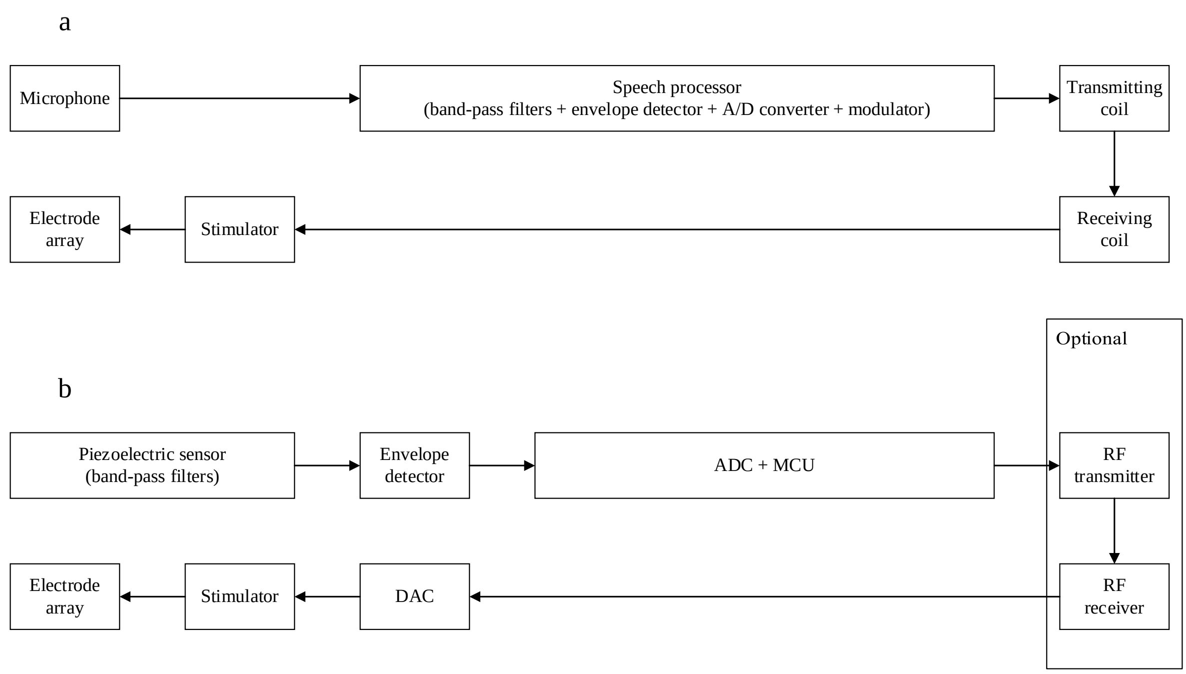

Figure 11. (a) Conventional (b) Entirely implantable cochlear system. The block diagram shows the components of two system. In the external unit, a speech processor is programmed with the desired signal processing algorithm to make a charge-balanced biphasic pulse from the incoming acoustic signal. On the other hand, the entirely implantable cochlear system uses the piezoelectric sensor to replace the band-pass filters. In this system, RF part is optional.

4.2 MATLAB Simulation

The MATLAB simulation is useful to verify the block diagram of system. In this simulation, the sound wav file is used as a sample for the output from the microphone. The acoustic sound is band-pass filtered by one of the 16 channels. The waveform of each block is shown in Figure 12 and 13. As shown in Figure 14, the biphasic pulses are continuously interleaved.

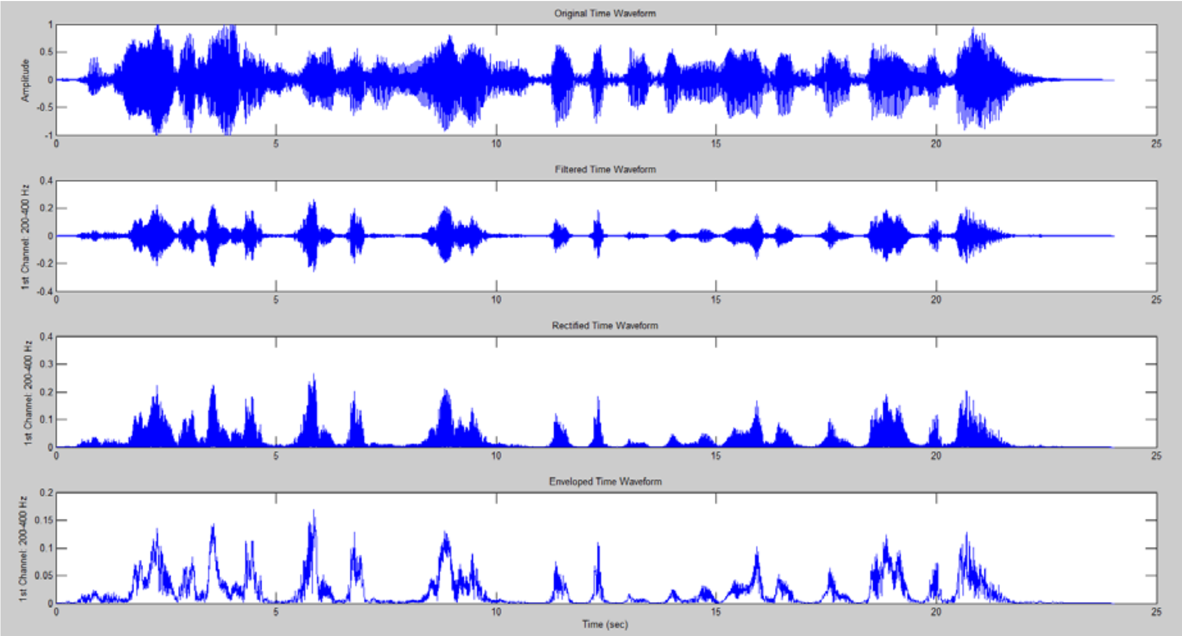

Figure 12. 1st channel waveform. In the 1st channel, the band is 200-400Hz. The original time waveform is band-pass filtered, rectified, and enveloped by 1st channel.

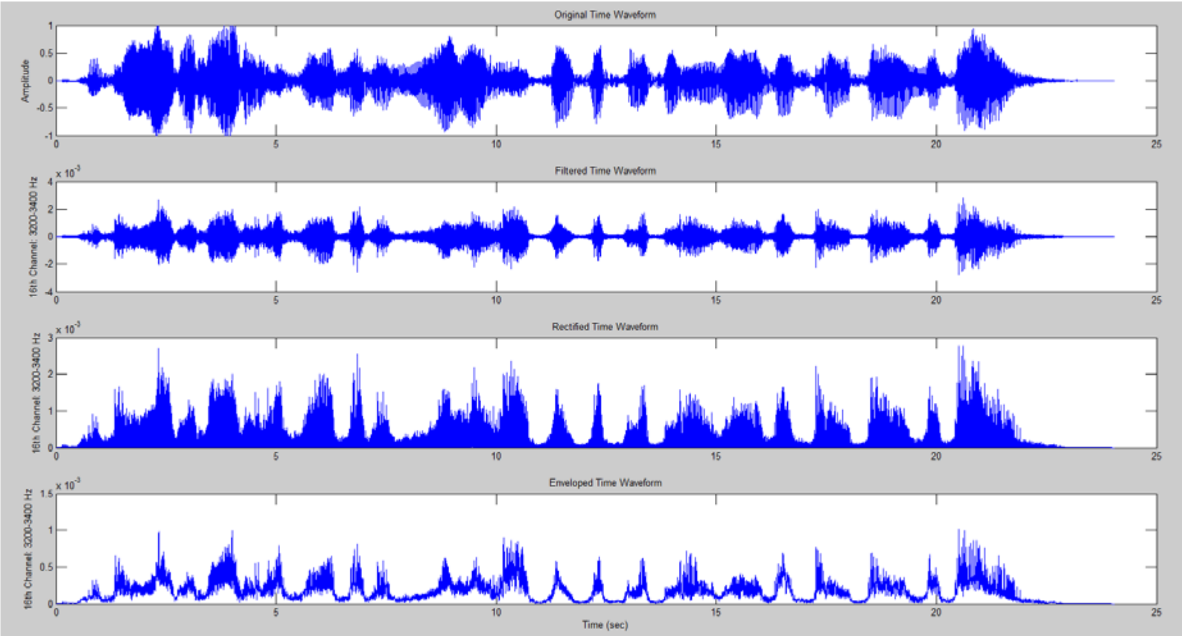

Figure 13. 16th channel waveform. In the 16th channel, the band is 3200-3400Hz. The original time waveform is band-pass filtered, rectified, and enveloped by 16th channel.

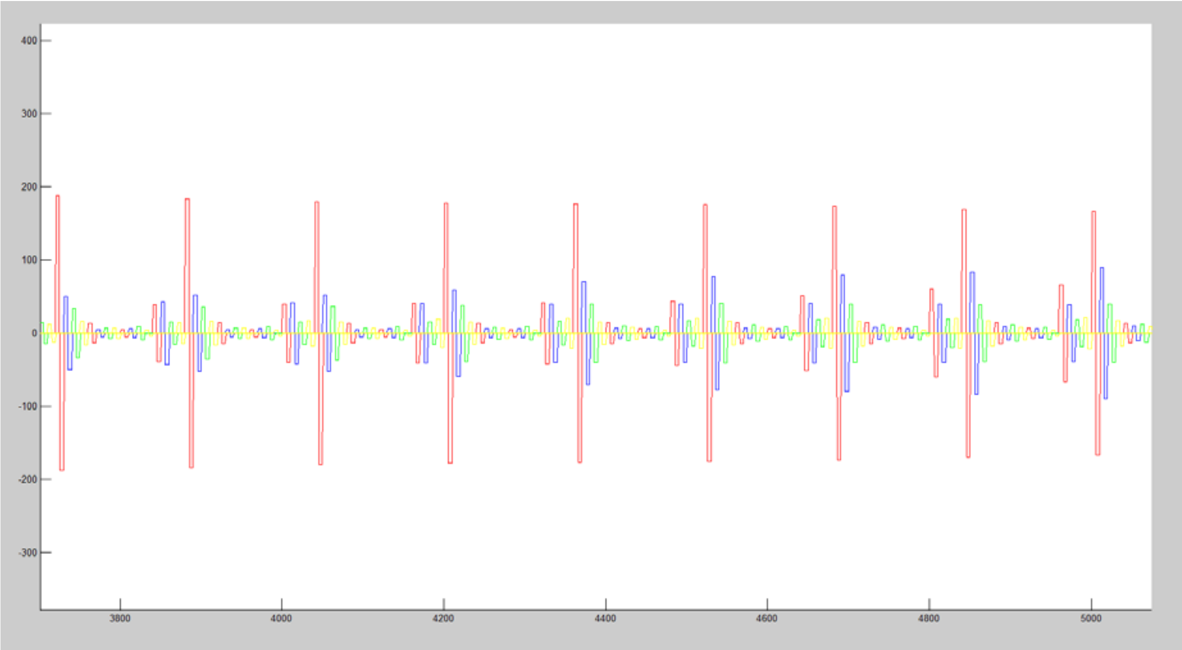

Figure 14. Biphasic pulse waveform. Non-overlapping biphasic pulses are continuously interleaved. The amplitude is affected by enveloped detector's output while the width is fixed.

4.3 System Development

The signal processing system is referred to as miniaturized system comprised of sensing, processing, and actuating functions which are combined onto a single board. The motivation of development is the need to realize functionality while reducing system size and power consumption. Building a complete PCB involves several works, as these works collaborate not only hardware (H/W), but also software (S/W) design. Hardware is manufactured almost exclusive in digital components, but many of the desired peripheral components are not compatible. Moreover, the hardware design is associated with the microprocessor that supports the software for the system. The successful PCB design requires integration from the hardware design at the component selection. A variety of tools for the support of the design exists for the system development. In the thesis, the OrCAD Capture and the Code Composer Studio are used to develop the PCB.

4.3.1 Component Selection

When beginning a PCB design, we need to choose proper components first. In this board, small sized parts are preferable to fit many parts onto the board. For example, the 0402 sized part is very small. As shown in Figure 15, the most important component is the MCU. Other peripheral ones are listed as below.

- Low-pass filter (LPF) only allows low frequency signals from cut-off frequency. In this board, the filter is designed using resistor-capacitor (RC) circuit. This filter is made up of passive components such as 47k resistor and 100pF capacitor. The cut-off frequency is 33.9kHz.
- Operational Amplifier (Op-Amp) can be connected by external resistors or capacitors. In this board, OPA4130 is available in IC package of quad op-amps within one single chip. The op-amp is designed using inverting amplifier circuit. The gain is calculated by 20k resistor and 1k resistor. Therefore, the closed loop gain of this inverting amplifier circuit is given 20 (26dB).
- High-pass filter (HPF) only allows high frequency signals from cut-off frequency. In this board, the filter is designed using resistor-capacitor (RC) circuit. This filter is made up of passive components such as 1uF capacitor and 330k resistor. The cut-off frequency is 0.482kHz.
- Envelope detector is used to reduce the sampling frequency. If the acoustic sounds do not have any high frequency, the envelope detector circuit could be used. This circuit consists of a diode and a RC filter. The diode in the circuit rectifies the input signal, and the RC combination allow the output signal to follow the envelope of the input signal. The RC filter is made up of passive components such as 10uF capacitor and 1k resistor.

- CC430F6137 is ultra-low-power microcontroller system-on-chip with integrated RF transceiver. This chip features the 16-bit reduced instruction set computer (RISC) architecture and 16-bit registers. It also includes a 12-bit analog to digital converter (ADC).
- Digital to analog converter (DAC) is used at the end of a digital signal processing where analog signal is required. Because the biphasic pulse is one of the analog signals at the electrically neural stimulation. In this board, DAC7678 is available in IC package of octal-channel within one single chip.

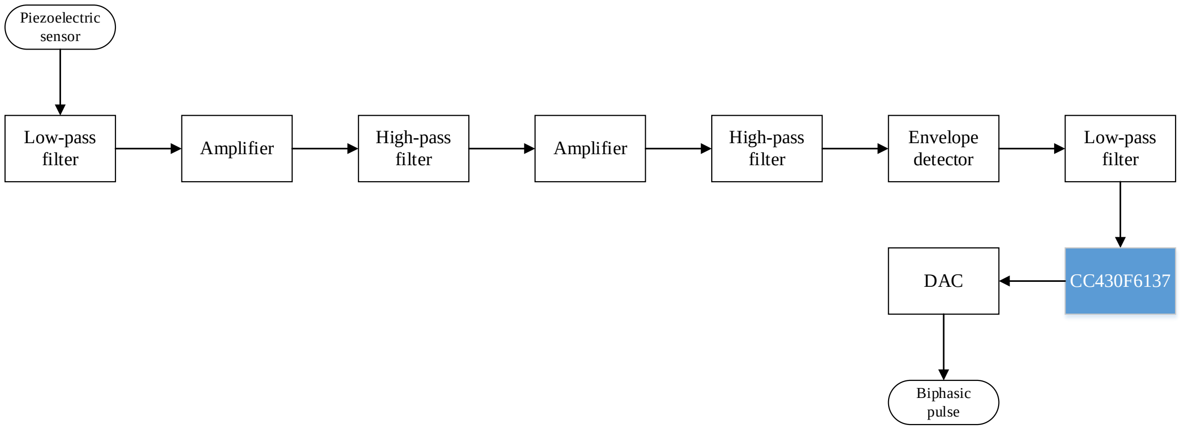

Figure 15. Components of the system. Within the scope of a practical design, the system is implemented on a Texas Instruments microprocessor. In the analog part, low-pass filter and high-pass filter are used to reduce board noise.

4.3.2 Hardware Design

The OrCAD Capture is well-suited to design the circuit for development of hardware. The detailed schematics are shown in Figure 16, 17, and 18. The schematic design is evolved from application notes, hardware manuals, and data sheets. The analog part amplifies the acoustic signals generated from the piezoelectric sensor. The signal is amplified through the OPA4130 which is designed using inverting amplifier circuit. Also, this part has low-pass filter and high-pass filter for reduction of board noise. The digital part consists of micro controller unit (MCU), digital-to-analog converter (DAC), and power management. In this part, the signals are converted into the biphasic signals in the MCU which is CC430F6137 of Texas Instruments. The result of signal processing can be shown in the DAC waveform which is the charge-balanced biphasic pulse. The power management supplies 3.3V in the circuit.

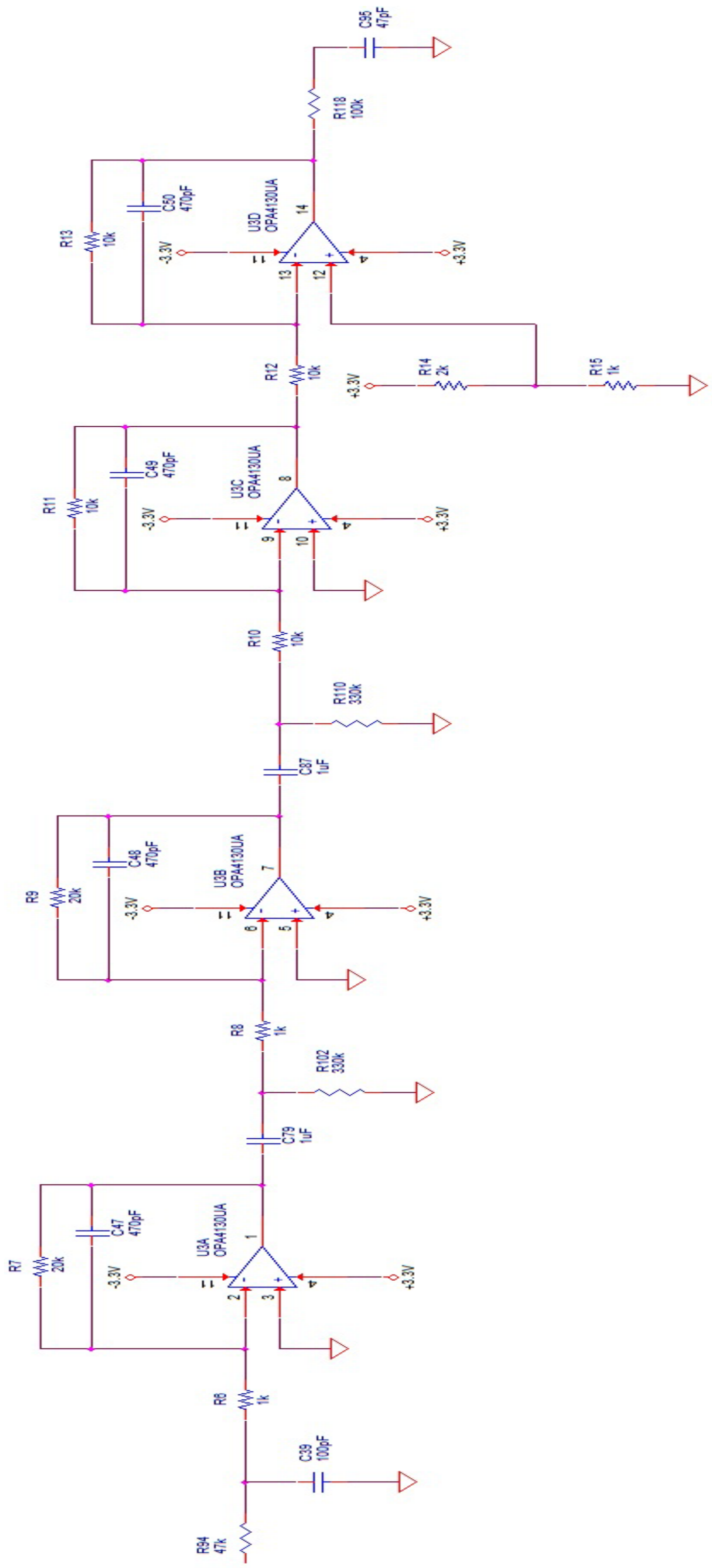

Figure 16. Schematic of analog part. This schematic is composed of low-pass filter, high-pass filter, inverting amplifier, and envelope detector.

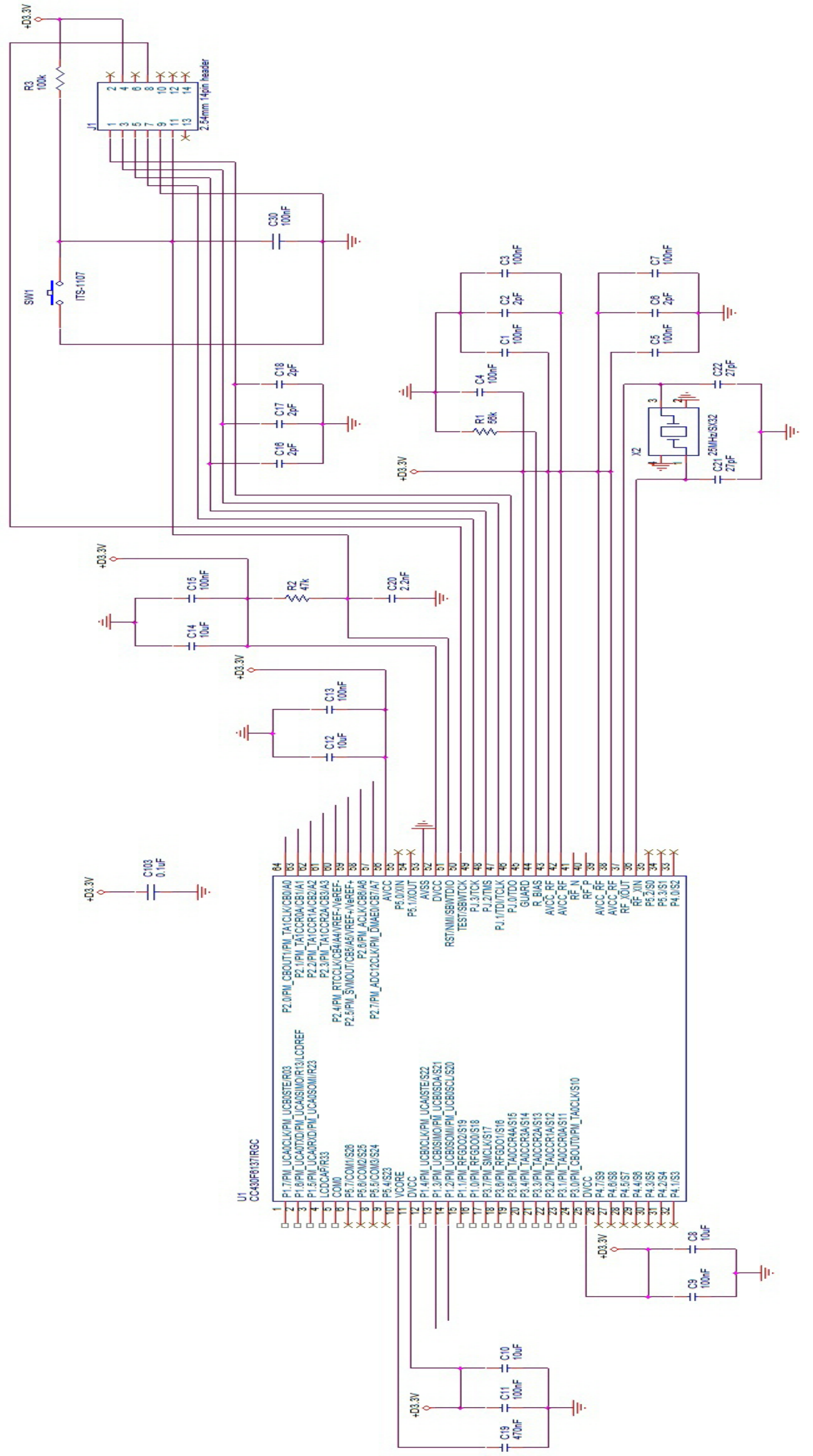

Figure 17. Schematic of digital part. This schematic is a reference design from the CC430F6137 based on the datasheets.

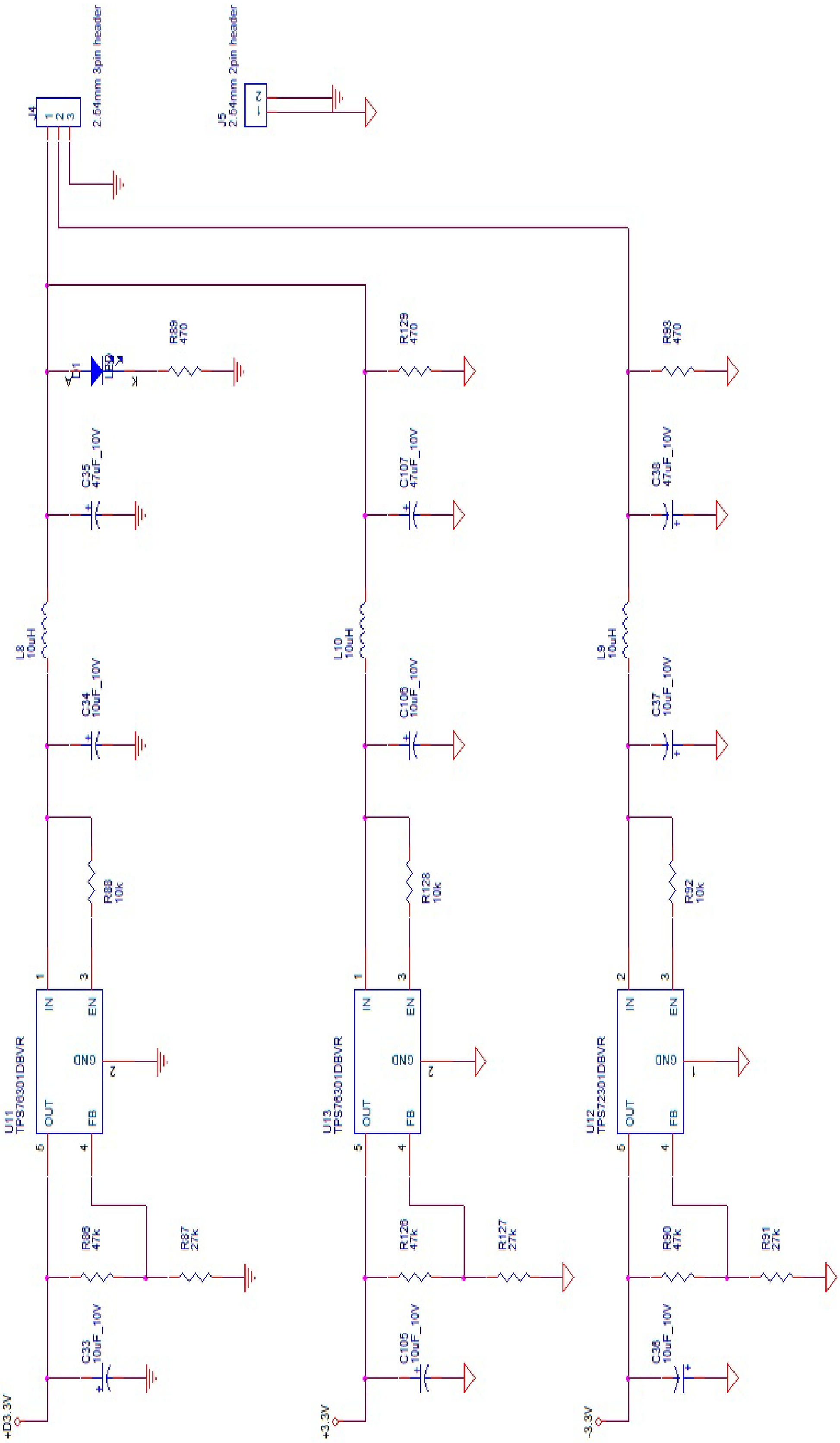

Figure 18. Schematic of power management part. The schematic is designed to supply ±3.3V voltage.

4.3.3 Software Design

The Code Composer Studio is freely available development tool with its microprocessor. The tool is an integrated development environment (IDE) for the Windows operating system with integrated C compiler and simulator. Programming can either be done in C or assembler. This software is written in C and has been integrated into a common workspace. It consists of a common library and project. To create a new project for a design, it is to reference an existing project and example. The software codes are written by example codes for CC430F6137. The structure of software code is shown in Figure 19.

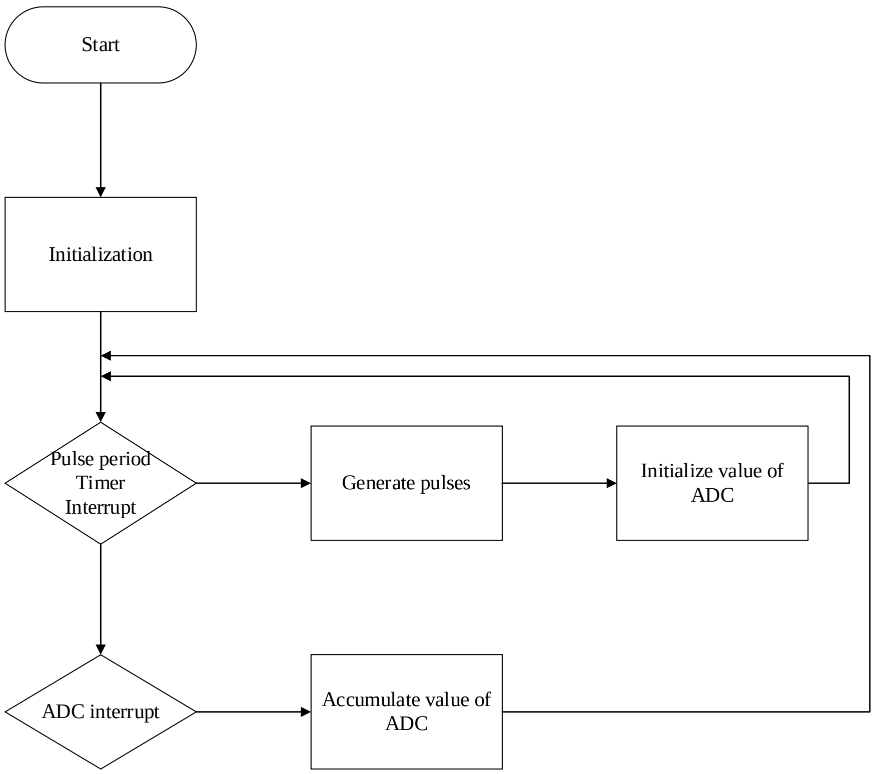

Figure 19. Software architecture. This software code is written by Timer_A and ADC10_A. Timer_A is a 16-bit counter with up to seven compare registers. It can support multiple compares, PWM outputs, and interval timing. It also has extensive interrupt capabilities. And ADC10_A module supports fast 10-bit analog-to-digital conversions.

4.4 Result of Printed Circuit Board (PCB)

The signal processing system was assembled on the PCB, shown in Figure 20, to verify the animal experiment. While the board is operated on normal mode, the size is not small. However, to achieve integration and be able to handle components easily, the surface-mount devices (SMD) parts of size 0402 (1.0mm * 0.5mm) are used. The pin header with 14 pins for JTAG downloader and the reset switch are on the board. They provide a one to one mapping of the pins of the CC430F6137. The three pin header is also assembled to provide power supply for the board. The eight pin header is connected by DAC. Lastly, the piezoelectric sensor is assembled by wire-bonding. When the PCB comes back from the foundry, it should be checked to ensure that all of the components and connections are working within normal parameters. The main features are as follows:

- The board is controlled by CC430F6137 with RF transceiver.
- Low power consumption (160uA at active mode, 2uA at stand-by mode, 1uA at off mode)
- Using piezoelectric sensor
- Size: 80mm (length) * 80mm (width) * 1.6mm (thickness)

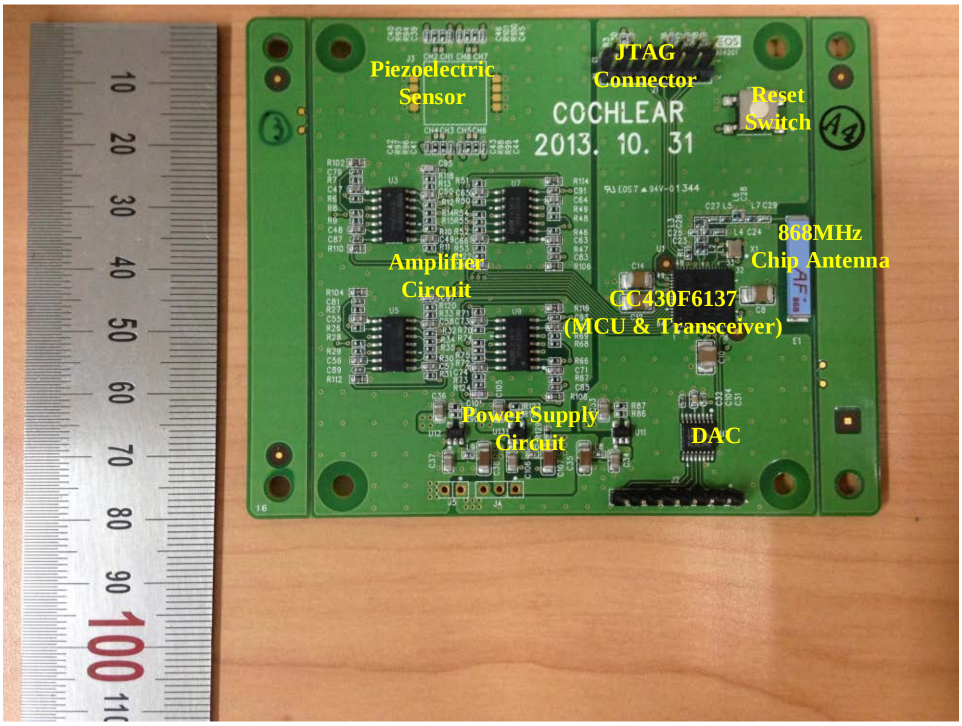

Figure 20. Result of the PCB. On the upper left of the PCB, a piezoelectric sensor is placed. Below the sensor, amplifier circuits are placed. On the right side, CC430F6137 and antenna are located. On the lower right side, DAC is placed. And power supply circuit and JTAG connector are located around the PCB.

4.5 Experiment of Signal Processing Board

The signal processing board is designed to provide the biphasic pulse at the electrode array with an auditory nerve. The software algorithm is developed to extract the waveform data that appear as a charge-balanced biphasic pulse in the voltage output. The results are measured by signal generator, power supply, and oscilloscope equipment. Figure 21 shows the single-channel output data. The system is designed to support the multi-channel for the recognition of better hearing. The multi-channel output data is shown in Figure 22. This output is shifted by delay scheme. This interleaved output waveform can be used to stimulate the auditory nerves at the cochlea.

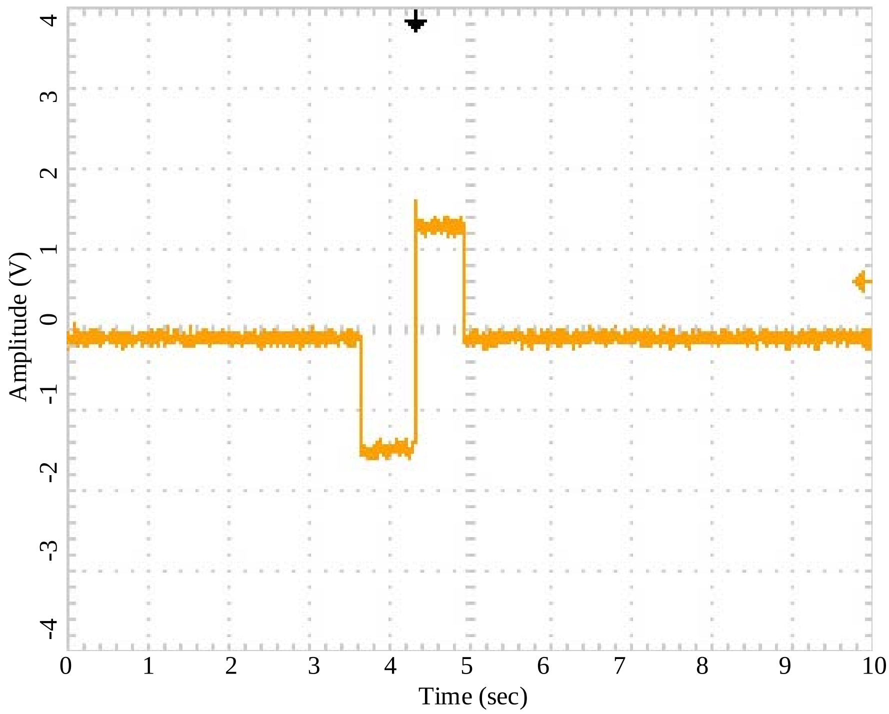

Figure 21. Biphasic pulse (Ch. 1). In the board, input signal is injected by signal generator. The voltage output waveform is measured by oscilloscope. This pulse can be shown in single channel mode.

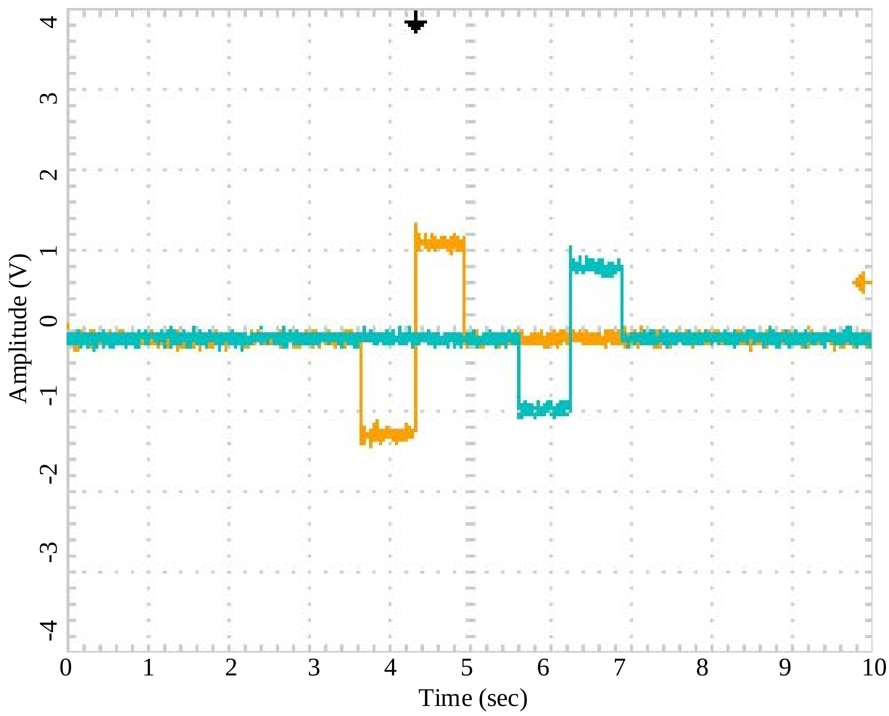

Figure 22. Biphasic pulses (Ch. 1 & Ch. 2). These pulses can be shown in dual channel mode. This board was designed to non-overlap the pulses.

Ⅴ. CONCLUSION & FUTURE WORK

In the thesis, main contribution is the design of signal processing using the piezoelectric sensor for the entirely implantable cochlear system. The goal of signal processing is to detect the sound channel by channel and to generate the biphasic pulse at the corresponding channel. In this board, the CIS algorithm is used to make the biphasic pulses. The signal processing algorithm is composed of band-pass filtering, envelope detection, and integration. The band-pass filtering is realized by frequency selectivity of piezoelectric sensor. The envelope detection is performed by analog circuit such as a diode and RC components. And integration is achieved by digital circuit such as MCU which is CC430F6137. The MCU is programmed by S/W source code. Even though the PCB is not implantable size, the biphasic pulses is measured by experiment setting. The biphasic pulses is shown to be generated in non-overlapping mode. In the future work, the signal processing board will be used to estimate the performance at the animal experiment.

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KOREAN ABSTRACT (국문요약)

압전센서를 사용한 인공와우시스템 신호처리설계

감각신경성 난청질환은 인간의 내이에 있는 청각세포가 손상되어 발생되는 증상으로써 현재 가장 적절한 치료법으로는 인공와우 시술이 있다. 인공와우는 청각신경에 직접 전기적 자극을 주는 방식으로 난청을 해결한다. 이와 같은 인공와우 시스템은 외부의 소리를 전기적 자극의 형태로 변환해주는 외부 신호처리부가 핵심이다. 하지만 인공와우를 사용하고 있는 환자들은 외부 신호처리부가 쉽게 노출되는 것을 꺼리며, 자주 배터리를 교체하는 것에 불편함을 느낀다.

인공와우 연구에서 외부 신호처리부를 작고 저전력으로 동작하게 하려는 연구가 활발히 진행 중이다. 최근 압전소자를 활용한 인공와우 시스템이 제안되었으며, 본 연구에서도 압전센서를 사용한 완전삽입형 인공와우 시스템을 다루었다. 제안된 시스템에서 압전효과는 외부의 물리적 힘에 의해 압전소자가 기계적 변형을 일으키면서 전기적 신호를 발생시키는 원리이다. 이와 같은 압전소자는 기존 시스템에서 많은 비중을 차지하고 있는 마이크로폰 및 외부 신호처리부의 소형화뿐만 아니라 저전력동작을 가능하게 한다.

본 연구에서 사용된 압전센서는 미세전자기계시스템 (micro electro mechanical systems, MEMS)을 통해 제작되며, 인공와우 신호처리에서 주파수 분리 기능을 수행한다. 따라서 완전삽입형 인공와우 시스템에서 신호처리는 압전센서의 주파수 분리 기능을 제외한 나머지 역할을 처리한다. 현재 제작된

압전센서는 주파수 대역별로 여러 채널이 존재하며, 해당 대역의 신호가 인가되면 특정 빔이 반응하여 전기적 신호를 발생시킨다. 이에 신호처리부는 이 신호를 사용하여 청각신경이 자극 받을 수 있는 형태로 변환해주는 것이 목적이다.

본 연구의 목표는 기존 마이크로폰의 역할을 대체하는 압전센서를 사용한 신호처리부의 구현이며, 향후 동물 청각인지 실험에서 사용 될 프로토타입 (prototype) 형태의 신호처리 보드의 제작이다. 개발된 신호처리 보드는 다 채널 시스템에서 성능이 입증된 CIS (continuous interleaved sampling) 알고리즘을 적용하였다. 다만 기존의 CIS 알고리즘과 비교하였을 때, 본 시스템에서는 압전센서가 대역 통과 필터 역할을 수행하는 차이점이 있다. 결론적으로 완전삽입형 인공와우 시스템에서 압전센서를 적용한 현재의 보드는 신호처리의 기능적 구현이었다. 향후 제작될 신호처리 보드에서는 ASIC (application specific integrated circuit)을 적용하여 현재 시스템의 크기 및 전력소모량을 개선시킬 것이다.

핵심어: 인공와우, 압전센서, 신호처리, 소형, 저전력

ACKNOWLEDGEMENT