Frequency Shifting for Patients with High Frequency Hearing Loss

(Using Digital Signal Processing Chips from Texas Instruments)

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Abstract

Digital Signal Processing is the study of the digital representation of an analog signal and the conversion and processing of those signals. There are many fields in which digital signal processing is becoming important, such as for communications, medical instrumentation, video, and audio. Our group will use digital signal processing chips from Texas Instruments to create an improved hearing device for patients with sensorineural hearing loss (also known as high frequency hearing loss). This semester, we plan on building the hardware to control input and output, and learning the software kit and programming the actual DSP chip. For input, we will use an electrot condenser microphone for its tiny size, the DSP chip we have selected, the C5509, is designed for use with audio signals and has a built in ADC. To compact the entire human hearing range into the limited frequencies available to those with sensorineural hearing loss, we are looking into using phase shifting, phase vocording, and resampling. In future semesters, we will attempt to streamline the code, reduce background noise, shift voices into the hearing fovea, and compact our device to look like a Bluetooth headset.

Background

Types of Hearing Loss

Our sense of hearing is only possible due to a complex set of organs, all with a specific and necessary purpose. Because there is so much that can go wrong in the ear with so many different parts, doctors have divided hearing impairment and loss into two broad categories: conductive and sensorineural hearing loss.

Conductive

Damage or obstruction to either the outer or middle ear is classified as conduction hearing loss. Obstructing the outer ear canal with either wax or a foreign object is one of the most common causes of conductive hearing loss, but damage or loss of the Pinna is also possible. The other major source of conductive hearing loss is infection in the middle ear. Infections occur in the Eustachian tube (which is narrow and more curved in babies) that will lead to a wax build up in the middle ear, and will dampen middle ear bone vibrations. Other cause of conductive hearing loss include an abnormal pinna, irregularly curved ear canal, scratches or scars of the ear drum, middle ear bone damage, or tight middle ear bone muscles.

Sensorineural

Sensorineural hearing loss is classified as pretty much anything else that can go wrong. Any damage to the cochlea, auditory nerve, or temporal lobe of the brain may cause hearing loss. Any lesions in the auditory nerve can destroy hearing in both ears, or hearing in just one ear, or certain bands of frequencies with no specific preference. There are many areas of the brain that cause also be harmed and this can lead to a multitude of other symptoms, too many to list them all here. However, the most common type of sensorineural hearing loss is damage to the inner hair cells of the cochlea.

Inner hair cells line the entire cochlea from base to apex, and the curling structure is designed to innervate certain regions of its length for certain frequencies. When the hair cells of one stretch of cochlea become damaged, signals for this frequency range are no longer sent to the brain and are, therefore, not observed. High frequency waves of the same amplitude as a lower frequency wave have more energy. Ergo, the high frequency hair cells are more likely to be damaged if the waves that innervate them are of greater energy. For this reason, high frequency hearing loss is the most common. In fact, our high frequency hearing begins to decrease after we pass 20 years of age. However, the frequency ranges that are used to detect speech will not be affected until, at the very earliest, the mid 70's (if completely natural). With the loud sounds and cheap headphones used today this process is natural process is being rapidly accelerated.

Prevalence of Hearing Loss

Americans don't seem to realize how valuable their hearing is until it is gone. Because of this attitude we continue to ignore the advice of our elders as they yell at us to, "Turn that loud noise down!" It is estimated that currently ten million Americans have an irreversible hearing disorder caused by excessive noise. Also, 30 million Americans are exposed to potentially damaging sound levels every day (NIDCD website, 2007). For many reasons, the number of Americans with a hearing disorder has doubled over the past 30 years. From 13.2 million in 1971 (Ries, 1994) to an estimated 28.6 million in

2000 (Benson & Marano, 1995). Furthermore, the most recent study estimates that 31.5 million people in the United States currently have a hearing disorder (Kochkin, 2005), that is nearly 10% of the population.

The incidence of hearing loss increases per age group from 17 per 1,000 for those under 18, about 71 per 1,000 for those between 29 and 40, and 167 per 1,000 for those between 41 and 59, up to 314 in 1,000 for those above 65, and up to 50 percent for those older than 75 years of age. (NIDCD website, 2007; Kochkin, 2005) However, this is not an affliction that only affects the elderly. As Figure 1 shows, over 65% of those with a hearing disorder are under the age of 65.

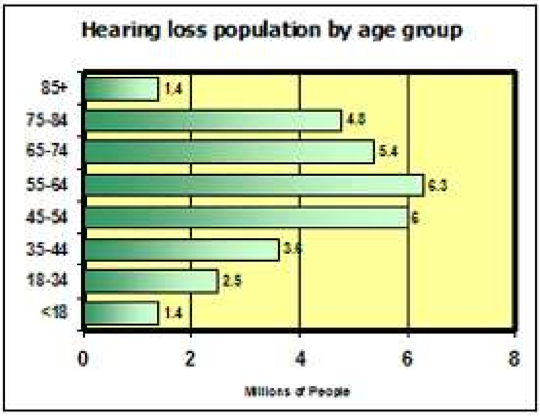

Figure 1. Hearing loss is not only an elderly condition. This graph shows the number of Americans with hearing loss in each age range. (Better Hearing Institute website)

It is most important to treat the hearing loss of the young that are just developing language skills. This is because a hearing deficit will cause children more difficulty when learning vocabulary, grammar, word order, idiomatic expressions, and other verbal skills (NICCYD, 2004). The Department of Education counted 5,775,722 with a known hearing loss attending public schools; of these only 70,767 (1.2%) received help for their disability (DoE, 2002). Studies show that up to 6 in 1,000 newborns are affected by a hearing disorder, and these studies admit the difficulties in detecting hearing loss in those without language skills (Cunningham, 2003). Those under 18 also make up over 8% of those with severe or profound hearing loss (Blanchfield, 2001).

Problem Statement

Our client, Texas Instruments (TI), has given us the task of engineering their digital signal processing (DSP) chips to work in medical applications. This project will give the team the opportunity to use and learn from TI's DSP chips in order apply them to medical applications. We have selected to use a DSP in implementing a hearing aid that can compensate for sensorineural hearing loss. As mentioned earlier, millions of people suffer from sensorineural hearing loss; however, the intensity of the hearing loss can vary from person to person. The normal human hearing range is from 10 – 20 kHz with normal speech in the range up to 10 kHz (Figure 2).

Our objective is to design a device that can shift frequencies that are beyond the wearer's hearing range into a frequency range that they can hear. Using DSP technology, will give us enough processing power to be able to shift frequencies in real-time and eliminate lag-time due to processing. Since shifting frequencies can produce some error and noise, especially shifting long distances, we will begin work on the normal human speech range, giving us a smaller

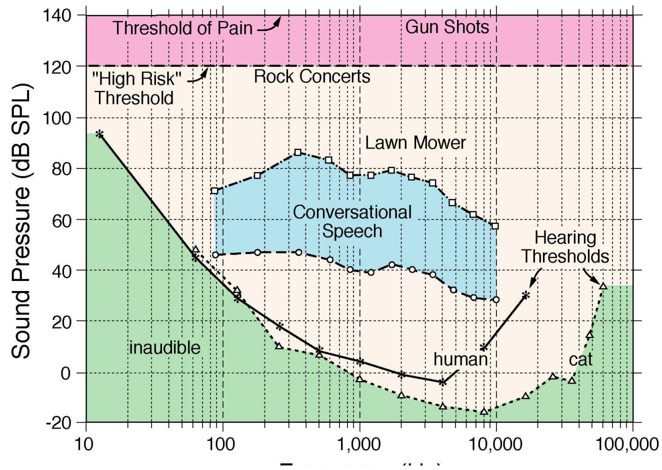

Figure 2. This graph shows the full range of what an average human can hear before any hearing disorders. The blue band indicates the range of normal speech. This will be our targeted region to begin our processing.

range to deal with and reduce the chance of electrical noise.

Since sensorineural hearing loss affects different people in varying degrees, the device will need to be calibrated to each user. We plan to develop a frequency scan to let the device calibrate itself to the user. The user will push a button on the device to initiate the frequency scan, which will produce a signal that progressively goes from a low frequency to high. The user will then push another button to indicate when they can begin to hear the signal and once more we they are no longer able to hear the signal. This will allow the device to identify the audible range of the wearer. Any sound outside the this frequency range will be processed and shifted into the wearer's audible range, allowing them to hear what they, by themselves, could not.

We have selected to use TI's C5509 DSP chip. According to TI, the C55 line of chips is power-efficient DSP, which are ideally designed for portable devices such as cell phones, PDA's, and MP3 players (TI website, 2008). These DSP's have ultra-low standby power, which will allow us to minimize the size of the device and maximize the usage life, as we ultimately will be able to supply power with a small battery. Because this device is meant to be worn, it is important to take aesthetical considerations into account. Not only does the device need to work, but also it shouldn't attract negative attention to itself. We would like to make it minimalist or fashionable, and ultimately hope to be able to make the device small enough to resemble a Bluetooth hands-free device for cell phones.

Design

Platform

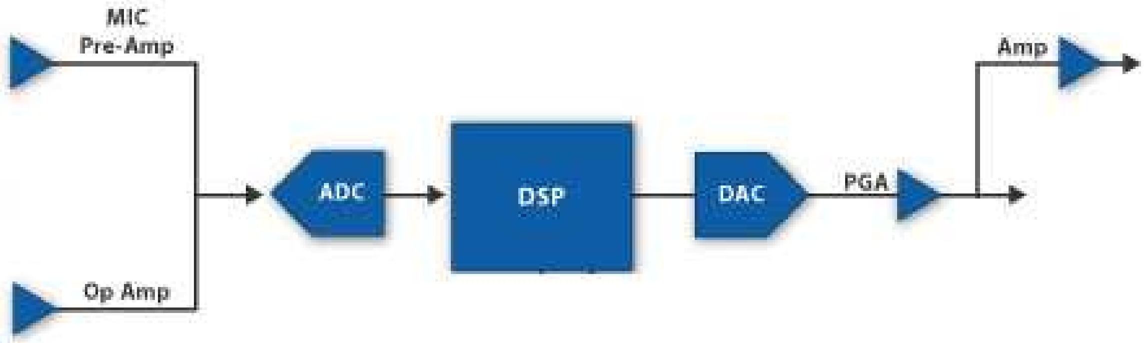

Figure 3. A block diagram for the design of the hardware, or platform, of our digital hearing aid.

A platform was selected based on the project requirements. The original signal is received by the microphone which will be connected to the DSP chip. Because this specific DSP has an analog to digital converter in the chip, there is no need to use an analog to digital converter before the DSP. The analog signal is passed into the DSP chip where it is converted to a digital signal and the necessary operations are performed on the signal (for example, compressing or shifting the frequency so that all the speech frequency will be in hearing range that a person with sensorineural hearing loss.) After the signal modification in the DSP, the processed digital signal is converted to an analog signal using a digital to analog signal converter, which will send the signal to an output, such as a speaker.

Signal Processing

Audio signal modifications could be achieved with several audio signal processing methods. In order to fulfill this device requirement, it is necessary to operate audio timescale-pitch modification. There are several ways to perform this operation such as resampling, phase vocorder, or pitch scaling. The method of using time domain and frequency domain duality could be the first method used. By performing a Fourier transform function the signal is converted from the time domain to the frequency domain. The type of transformations that the signal will undergo is easier in the frequency domain. A signal in the frequency domain could be shifted or compressed in the frequency domain so that the all the speech frequency range could fit within the frequency range of a person with sensorineural hearing loss. Shifting a signal in the frequency domain would not cause expansion in the time domain; however, it would result in an overlap of certain frequencies. On the other hand, compressing within the frequency domain would not result in an overlap of frequencies; it would lead to an expansion of the signal in the time domain. This expansion could be compensated by resampling the frequency signal during the calculation of the inverse transfer function. Because calculation of frequency domain signals requires a certain amount of sampling in the time domain, an output of modified signal would result with some delay. However, this delay can be ignored because the DSP would have a high sampling rate.

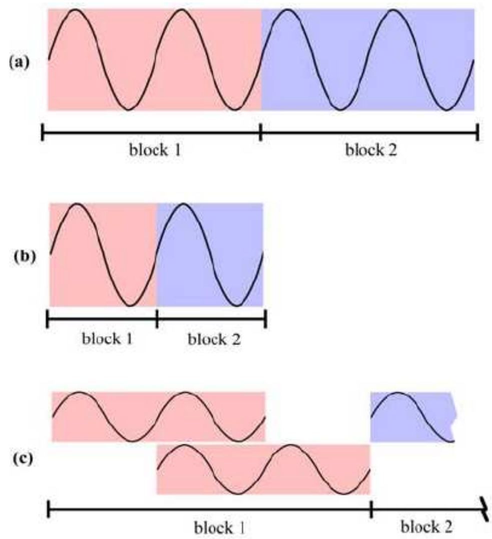

Figure 4. Block-based time domain pitch shifting technique; (a) shows input signal split into blocks; (b) for a down shift, input blocks are truncated by an integer number of periods to create a signal of shorter duration; (c) for an up shift, input blocks are overlapped and added to increase signal duration (from http://www56.homepage.villanova.edu/)

One other way to

perform this audio frequency modification is by pitch scaling (pitch shifting, or transposition.) This method could move the frequency without changing the tempo of a signal. Figure 4 depicts how this operation can be done in the time domain. Because this method does not require calculation of frequency domain signal, this method can modify a signal faster than using the method suggested above (the method using frequency domain to modify time domain signal).

Future work

This semester

The main goal of this semester is to build a platform and experiment with coding of the DSP chip. We are going to order the development kit from TI and study how to modify the DSP chip. At this moment we do not know what program is used to modify the chip. Before we get the chip we will perform trials of our coding techniques in LabVIEW or using MATLAB. Once the kit is received, we will program it from trials with other software. At this stage, hardware other than the chip—such as the microphone and speaker—are not needed as we can always input a signal from a DMM and study its output through LabVIEW. The microphone and speaker will be integrated into the circuit after the chip is successfully programmed.

Testing

Testing will be performed using LabVIEW. We can input signals of different frequencies and study the real time response from our code. Length of the code needs to be considered as it will greatly affect the efficiency of the chip. We may first create code that just gets the job done, and will look into trimming it up and streamlining it once we know it is effective.

Future semesters

The whole platform is to be built including the circuit, sound receiving device (microphone), and speaker. Work on reducing background noise or increase the SNR has to be done. We will want to refine the code to make the processing as fast as possible, and will continue to conduct tests. Explore the possibility to combine recent hearing aid features such as noise reduction and feedback reduction in our device. The hearing aid can also be used to determine the person's hearing fovea and perform compression and shifting to put the most relevant hearing information in that band. Housing of the device should be small and if possible, good looking, such as a Bluetooth headset.

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