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HEARING AIDS BASED ON SMARTPHONE BY IMPLEMENTING TYMPANIC MEMBRANE AND BASILIAR MEMBRANE COMPRESSION

[1]RULLY SOELAIMAN, [2]ARIANTO WIBOWO

[1,2]Department of Information Technology, Tenth November Institute of Technology, Surabaya, Indonesia

*Corresponding Author

E-mail: [1]rully130270@gmail.com, [2]louis.arianto@gmail.com

ABSTRACT

Hearing Impairment is one of the health problems usually caused by age or exposure to loud sound continuously. Based on World Health Organization, around 360 million people worldwide suffer from hearing loss and 32 million of these are children. Failure of the sound signal to reach the brain is the main cause of hearing loss. Hearing impaired often cannot hear soft sound, cannot hear in noise, and cannot hear sound in certain frequency. One of the solution is to wear hearing aids on the impaired ear. However, hearing aids are pretty expensive and some people cannot afford it. Without hearing aids, hearing impaired won't be able to understand daily conversation clearly, thus reduce the individual's ability to communicate with others and their hearing organ will continue to deteriorate. Unaddressed hearing loss also have effect on academic performance of children. Hearing Aids based on smartphone is an inexpensive solution for hearing impairments. Hearing impaired can download the applications in their smartphone so they can use the smartphone as the hearing aids. The application will mimics how ear works by implementing tympanic membrane and basiliar membrane compression. This paper intend to build a hearing aids application on smartphone using algorithm designed to mimic how the human ear works. Application will receive sound input from the microphone and output sound with better quality to the earphone of the hearing impaired. The system have been tested to hearing impaired with different environment (quiet and crowded environment), and they can listen to 79.3% of the given sound. With this satisfying result, we hope that this application will help hearing impaired to hear conversation clearly without the need to buy hearing aids.

Keywords: *Hearing, Hearing Impairment, Hearing Aids, Smartphone, Disability*

1. INTRODUCTION

Assistive Technology is a technology to help people with disabilities. With the development of information technology recently, assistive technology can help disabilities to do things which couldn't be done perfectly before in their daily life [1].

One of the disabilities often experienced by human is hearing loss. Over 5% of the world's population are suffering from hearing loss, which is around 360 million people worldwide including 32 million children. The majority of these people are from low and middle income country. Hearing impaired usually having a hard time to listen to quiet sound, listen in crowded enviroment, and capturing sounds in a certain frequency. This can affect individuals ability to communicate with others as they cant understand the context of the speech clearly. Exclusion from communication can have significant impact on everyday life, such as feeling of loneliness, isolation, and frustration [2]. Hearing loss can be caused by several factors, including genetics, ageing (degeneration of sensory cells), ear infections, excessive noise, etc [3].

Hearing loss can be treated using body language. One of the most widely used methods for people with hearing loss to understand the speaker is by reading the speaker's lips [2]. Thus, they can understand the context of the conversation. However, there are weaknesses in this methods, ie the patient cannot watch movies without subtitles, making phone calls, and can't communicate in dark environment.

Another way of dealing with hearing loss is to use hearing aids. These instrument receives sound from environment as the input, then performs processes to improve the sound quality. Then, the instrument emits the processed sound in accordance with the hearing losses characteristics of the user.

Through hearing aids, hearing impaired can listen to speakers, movies, and in dark environment [4]. However, the prices of hearing aids are quite expensive and less accessible to low-middle income people [2]. The prices can range from several hundreds to several thousands of dollars each. The prices varied according to the needs of the hearing impaired. Seeing the prices for each hearing aids (if both ears get damaged, patients will need a pair of hearing aids), this solution cannot be accessed by most people with hearing loss. This solution can only solve the problem of some hearing impaired.

For this purpose, we build hearing aids based on smartphone. In last decades, the development of smartphone is very rapid and most of the world's population already has at least a smartphone. We build smartphone's application that receives audio input from microphone, then performs sound quality improvement process on the smartphone based on how human's ear works. Improved sound will then be emitted by user's headset or earphone. These processes should be done in real-time to make the conversation feels natural. Hearing impaired won't need to buy hearing aids anymore as they can simply download the application on their own smartphone and use it to hear sound like normal hearing people.

Our objective is to build a smartphone based hearing aids which enhance the quality of input signal from microphone and output it to the earphone of hearing impaired. We hope that the system built can provide a practical solution to help hearing impaired to listen like normal hearing people. We will evaluate the functionality by testing the application to hearing impaired. We also evaluate the system built in this paper by comparing to similar system. System comparison will be based on 4 categories: sound quality, system flexibility, ease of use, and overall system. Both evaluation will use hearing impaired as participant.

The remaining paper is organized as follows. Section 2 presents a review of relevant studies. Section 3 explains proposed approach. Section 4 depicts evaluation of the application. Finally, section 5 shows conclusion and future research directions.

2. RELEVANT STUDIES

2.1 Acoustic Reflex Compression

Acoustic Reflex is a hearing mechanism done by stapedius muscle. Normally, sound will enter cochlear nucleus inside brain and the signal will be passed to superior olivary complex. After that, signal will be passed to facial nerve nucleus which have the control of stapedius muscle. If the sound is too loud, facial nerve nucleus will give orders to stapedius muscle to contract. This involuntary contraction by stapedius muscle will reduce the compliance of the sound that pass through tympanic membrane. This mechanism will occur in response of high intensity sound from environment or from the person's own vocal cords when the person is speaking. Loud sound received from left / right ear only will also cause the other ear's muscle to contract. This is due to cochlear nucleus will pass the signal to both superior olivary complex, left and right. Thus, both superior olivary complex will give orders to both facial nucleus to contract the stapedius muscle.

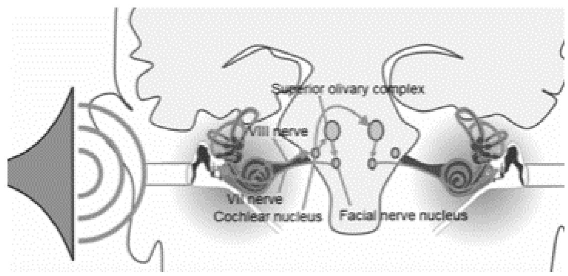

Figure 1: Acoustic Reflex Mechanism

Figure 1 explains how Acoustic Reflex works when given sound from left ear only. Notice that right ear's muscle is also impacted by stimulus given to left ear. Acoustic Reflex has a threshold (ART) in which sound with intensity higher than ART will trigger the acoustic reflex. These threshold are varies and people suffering hearing loss usually have higher thresholds [5]. As the process includes processing the sound in the brain, acoustic reflex have some delay before reducing sound through middle ear when loud sounds emerge.

This process aims to protect hearing organs from high intensity sound as hearing can be damaged when exposed to excessive noise [6]. Acoustic Reflex also reduce the intensity of sound from the individual's vocal cord. Individual's vocal cord is located near to ear drums and the sound received from vocal cord is loud when the individual is talking. By having acoustic reflex mechanism, we reduce our own voice while speaking so we still able to hear sound produced by surrounding environment.

2.2 Butterworth Filter

Butterworth Filter is one of the most used filter in signal processing to smooth the signal. This

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filter prototype as a low pass filter have a transfer function like below:

$$|H(j\omega)| = \frac{1}{\sqrt{1+(\frac{\omega}{\omega_c})^{2n}}} \quad (1)$$

ω_c is the cutoff frequency which frequency below will be passed and frequency above the cutoff will be attenuated. n is the filter's order. The more value of n, the cutoff will be sharper [7]. As n approaches infinity, butterworth filter will become ideal filter. The filter's response by tuning the order n can be seen in figure 2.

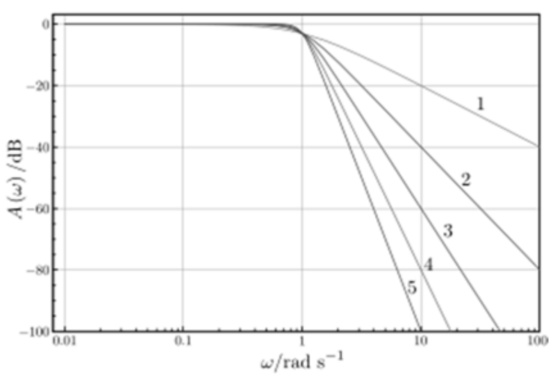

Figure 2: Butterworth Filter Orders

There are some advantages in using butterworth compared to other known filters, such as bessel and chebyshev filter. This filter works in all frequency range and has monotonically decreasing frequency response. This filter also doesn't produce ripple response near the passband [8]. The system will use butterworth filter bank, ie collection of butterworth band pass filter with different frequency range. These chosen filters frequency range mimic how hearing works and does not overlap with each other.

2.3 Basiliar Membrane Compression

Basiliar membrane is a long stiff element running inside the cochlea. Basiliar membrane has many section with different width, stiffness, mass, and dampness. Every section is sensitive to a frequency range, therefore its main purpose is to disperse signal's frequency. The base of the basiliar membrane is narrow and stiff compared to the apex, and is sensitive for high frequency while the apex is sensitive for low frequency.

Basiliar membrane also displace differently at different intensity [9]. This displacement affects how big the sound signal that enter the brain. Figure 3 shows the basiliar membrane displacement in response of different intensity stimulus.

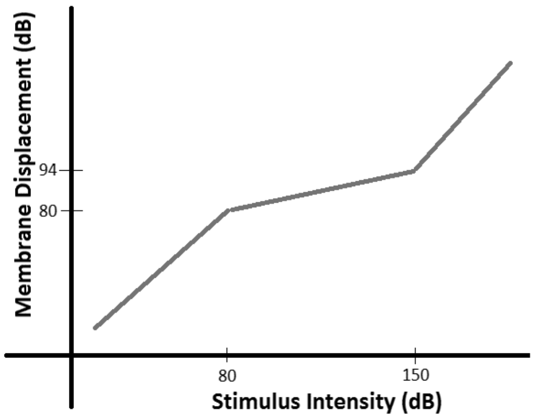

Figure 3: Basiliar Membrane Displacement

From the figure, we can see that Basiliar Membrane (BM) displacement respond linearly to the given stimulus in low intensity. Then, BM doesnt respond as much as the stimulus on higher intensity [10]. In this intensity range (80 decibel to 150 decibel), the input sound will be compressed (intensity reduction). Finally, BM will displace linearly again at very high intensity stimulus. Normally, BM respond linearly (first part of the graph) until 80 dB. After 80 dB, BM only respond at 0.2 dB for every 1 dB gain in the stimulus. This conclude that there's compression in the BM after 80 dB [11].

Compared to Acoustic Reflex Compression, BM compression is done after frequency dispersion, while acoustic reflex is done before (tympanic membrane and stapedius muscle are located before cochlea). Acoustic reflex also have a higher threshold than BM compression.

2.4 Noise Gate

Noise is unwanted signal in digital signal processing. Noise can be caused by electric circuit, digital quantization, or noise from the environment while recording. Noise can reduce human's understanding when presented with speech and also make us uncomfortable. Noise gate is a method that try to eliminate this noise from the sound as we only want to listen to the signal.

In noise gate, what differs signal and noise is the intensity. Signal tends to have high intensity, while noise usually have low intensity. This algorithm works by giving a threshold value. Sound signal which have intensity higher than the threshold will be considered as signal, while the one below threshold is the noise. The gate will close

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when noise is detected (intensity drop below threshold) to cut the noise from passing [12], thus the noise will be suppressed. The gate will open again as soon as signal is detected. This mechanism will remove noise in sound signal from passing the noise gate.

Figure 4 explains how noise gate works by passing signal above threshold and cutting all signal below threshold. To make the gate smoother, noise gate also have attack and release mechanism. Attack is delay given before signal is passed again, and release is delay given before the gate closes as the signal drops below the threshold. Attack's value is typically small as we want to hear speech as soon as possible, while release's value should be higher. Release ensure the gate doesn't close during short pause between words or sentences [12]. Therefore, release's value should have higher value to make people understand the context of the speech.

membrane compression, noise gate, and add gain to each channel so hearing impaired can set the gain according to their hearing loss characteristics. The system will have 9 butterworth filter bank. Basiliar membrane compression, noise gate, and channel gain will be implemented in each channel of the filter bank. Channel gain is implemented in each channel as hearing loss may have different characteristics in different frequency. Settings of each channel gain are available to set for users. We hope that user can choose their own settings which is best for their hearing loss. Acoustic reflex compression is implemented before frequency dispersion (butterworth filterbank) as in the hearing organ, tympanic membrane is located before cochlea where frequency dispersion occurs. The overall system design are showed in figure 5.

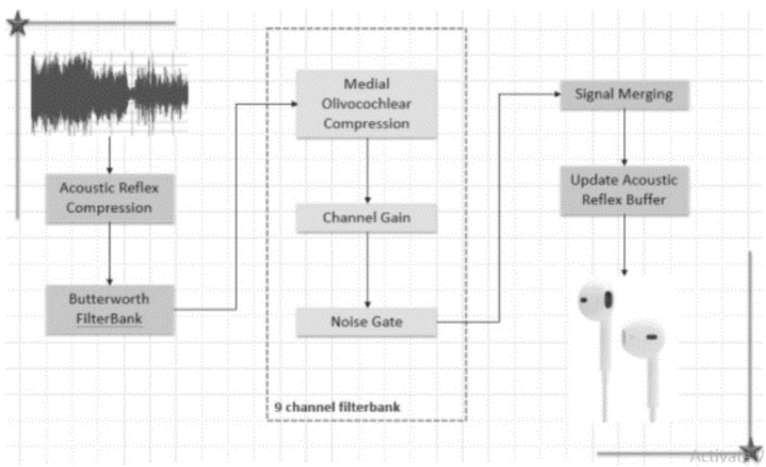

Figure 5: System Design

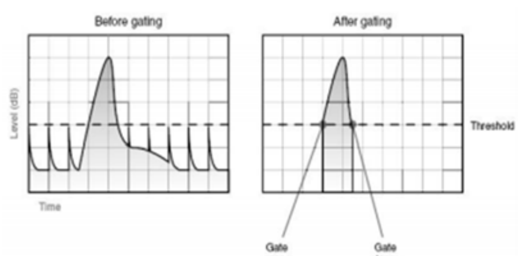

Figure 4: Noise Gate [12]

3. PROPOSED HEARING AIDS SYSTEM

Proposed system try to mimics how human ear works. The system consists of several module that aims to help hearing impaired to listen and understand speech clearly. We implement acoustic reflex compression, butterworth filter bank, basiliar

3.1 Input Signal

Input signal is taken from microphone with sample rate of 44100 Hertz. Nyquist-Shannon sampling theorem states that sample rate should be minimum 2 times maximum frequency that will be produced. As human can listen sound up to 20 kHz, so the minimum sample rate should be 40 kHz. As we want the system to be real-time, higher sample rate will cause the complexity higher and produce hissing noise, so 44100 kHz is selected. The input signal is mono (audio with single channel) as this input comes from microphone which only have one audio recorder.

3.2 Acoustic Reflex

Input signal will then undergo acoustic reflex compression. This compression aims to reduce the intensity of very loud sound. We implement a delay of 0.01 seconds before the compression takes effect. This means if loud sound is detected, it will need 0.01 s econds before reducing the input signal. Acoustic Reflex consists of 2 parts, acoustic reflex compression and update acoustic reflex buffer. The former part is reducing the intensity by a value which is computed in the latter part. The compression is done to the input signal, where the computing part (update acoustic reflex buffer) is done at the end, right before outputting the sound to the earphone.

Before we output the sound, we compute the acoustic reflex constant that will be used to reduce the intensity. The acoustic reflex constant can be calculated by this formula :

$$c = \frac{\sqrt{opf(in^2)}}{t} \tag{2}$$

in is the signal, *t* is the acoustic reflex threshold (ART) and *opf* is one-pole filter to smooth the signal. The value of *c* will be adjusted to 1 if the result of the calculation is less than 1 to prevent the sound to become louder. The one-pole filter choosen has the transfer function like below:

$$|H(z)| = \frac{b_0}{1 + a_1 * z^{-1}}$$
$$b_0 = 1/(sr * 0.06) \tag{3}$$
$$a_1 = (1/(sr * 0.06)) - 1$$

sr is the sample rate of the audio signal which is 44100 Hz. Acoustic reflex constant will then be inserted to the buffer (queue-like data structure). For the acoustic reflex compression itself, it will grab the constant from the buffer and reduce the intensity from the constant.

3.3 Butterworth Filter Bank

Butterworth filter bank is a collection of butterworth band pass filter. Each bandpass filter have different configuration of center frequency and bandwidth, and none of the frequency range are overlapping. We use butterworth filter of order 2 as the complexity can be done by smartphone's processor. This filter bank mimics cochlea which disperse frequency along the basiliar membrane. In the system, we build 9 butterworth band pass filters (9 channels) with each band pass filter configuration listed in Table 1.

Channel	Center Frequency (Hz)	Bandwidth (Centz)
1	250	600
2	500	600
3	1000	600
4	1414	300
5	2000	300
6	2828	300
7	4000	300
8	5656	300
9	8000	300

Table 1: Butterworth Filter Bank Configuration

There's no fixed frequency configuration for the filter bank. Other configuration may also do well. This frequency configuration is chosen as it represents frequency dispersion by cochlea. Centz is a dimensionless unit to measure frequency ratio. This frequency ratio represents the bandwidth better than Hertz as Centz is computed in logarithmic. Formula 4 shows how to convert Hertz to Centz bandwidth.

$$¢ = 1200 * log_2(f_2/f_1) \tag{4}$$

3.4 Basiliar Membrane Compression

Basiliar membrane compression is used to reduce the intensity of sound after threshold. In the system, we use 80 dB as the basiliar membrane threshold. 80 dB is the average basiliar membrane threshold of normal hearing people, and hearing impaired usually have similar threshold. If the signal's intensity is more than 80 dB, it will be compressed such that every gain of 1 dB, only 0.2 dB is passed from the basiliar membrane compression. The power can be calculated using formula below:

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$$out = in^{g} * t^{1-g} \tag{5}$$

in is the input signal, *g* is the gain after the threshold (0.2 in this case), and *t* is the basiliar membrane threshold (80 dB). The sound produced from this methods seems to be distorted for normal hearing people. However, it helps with hearing impaired to be able to understand the speech.

3.5 Channel Gain

Channel gain aims to increase the intensity of the sound in each channel so that hearing impaired may hear the sound like normal hearing people. The gain is applied to each channel, not all the signal at one cause some people may experience hearing loss at low frequency, but not at high frequency, and vice versa. Every hearing loss has its own characteristics and we supply it by allowing users (hearing impaired) to determine the gain for each channel themselves that best suits for their hearing loss [13]. Each channel will have minimum gain of 0 decibel (this means the signal will not be amplified) and maximum gain of 50 decibel. Amplification above 50 decibel can harm hearing organ, thus the system won't allow gain more than 50 decibel. User will be able to set the gain for each channel in the main and only page of the application themselves. There are some ways to tune the gain. The easiest way is to follow the audiogram each hearing impaired has. Audiogram is graphic representation of hearing ability resulting from hearing test performed by hearing health care professional. The main page and gain settings of the application can be seen in figure 6.

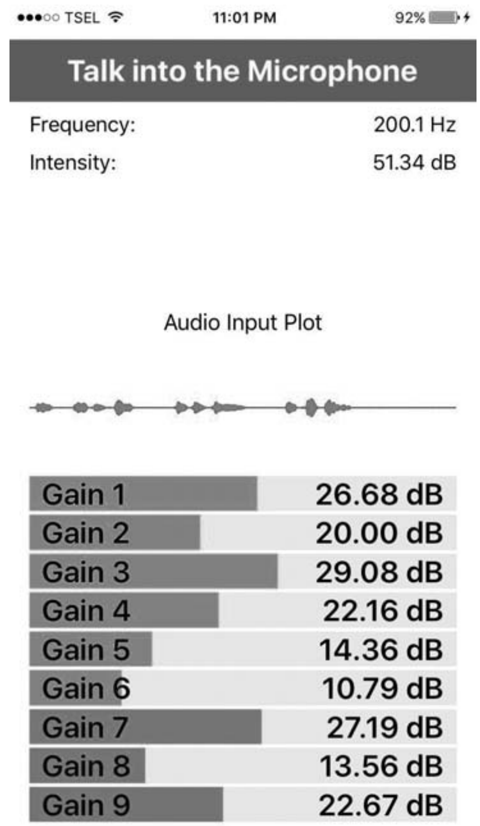

Figure 6: Channel Gain Settings

3.6 Noise Gate

We use 50 decibel in this system as the noise gate threshold, which mean any signal lower than 50 decibel is considered to be noise, and vice versa. This noise gate threshold is chosen by experimenting multiple times in quiet environment and crowded environment. Noise gate also have 2 other parameter, attack and release time. Attack time should be small as we want users to be able to listen to speech just as speech detected, so the attack time is chosen to be 0.01 seconds. Release time will be given a higher value of 0.3 seconds, so the gate will not open and close during short pause and help the listener to understand the speech context.

4. EXPERIMENTAL RESULT AND ANALYSIS

Experiment are conducted by trying the hearing aids application to hearing impaired. The participant of the experiment involved 7 people with different age, gender, and hearing aids usage duration. The variation of age, gender, and hearing aids usage duration hopefully can represent hearing loss in general. They are all suffering from hearing

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loss and therefore facing difficulties to communicate with others without hearing aids. However, their hearing loss characteristics are different with each other. There are 2 types of evalution done: functionality evaluation and comparison to similar system evaluation. Functionality test evaluates whether the system built can help hearing impaired to listen to surrounding sound. The comparison evaluation measures the performance of the system built compared to similar smartphone based hearing aids available at app store. We will use 2 hearing aids applications for this purpose: i-Hear and Petralex. Both of these evaluation will use the same participant.

During the functionality test, participants are asked to use the application and microphone. They will first tune the channel gain parameter for each channel that best suits for their hearing. The participant will choose their own settings based on their hearing characteristics, while we help them by having conversation and turning on television to find the perfect fit. The test is divided into 2 parts, the first part is hearing test in quiet environment, and the second is hearing test in crowded environment. In quiet environment, we evaluate the system by having conversation in an empty room. In crowded environment, we simulate it by turning on television, buzzing sound of electronics (eg fan), turning on videos in the background, or having background conversation that cause some noise in the background. Then, we will speak one word and asks the participant to repeat what we were saying. Accuracy then will be calculated by seeing how many words are repeated correctly divided by the number of words spoken. Speakers are located about 50 cm with the participants who try to hear the words. The words chosen for the experiments are words with low syllables (1 syllable words), i.e. cat, hands, bells, king, car, tree, book, chair, dog, and leg. We choose words instead of sentence in this experiment as human can guess words if they know the context of the speech. Using sentence to do hearing tests will cause an inaccurate result, as they may not hear the words well, but they guess it right as they understand the context of the speech. The same reason applied about why words with low syllables are choosen (in this tests, 1 syllable words). Table 2 lists our participants data.

Table 2: Participants Data

Participant Number	Gender	Age (years)	Hearing Aids Usage Duration (years)
1	Male	22	4
2	Female	89	9
3	Female	25	1
4	Female	57	-
5	Female	33	3
6	Male	38	4
7	Female	23	1

All of the participants will be asked to listen in quiet environment and crowded environment. In each environment, participants will be given 10 words as mentioned before. The experiment's result can be seen in Table 3.

Table 3: Functionality Experiment Result

Participant Number	Quiet Environment Accuracy	Crowded Environment Accuracy
1	100%	100%
2	50%	60%
3	90%	90%
4	70%	70%
5	100%	80%
6	80%	60%
7	80%	90%

Based on the experiment, the proposed hearing aids system based on smartphone can help hearing impaired to listen to speech in both quiet and crowded environment. One exception is to the second participant. Her accuracy seems to be low as she doesn't understand English. Other than that, the accuracy is adequate which means they can listen to the speech clearly. There's also a little accuracy difference between testing in quiet environment and crowded environment. Some of the participant even have higher accuracy in crowded environment. This makes us believe that background noise doesn't affect participant's hearing too much. In other words, the hearing aids proposed can be used in quiet and crowded environment.

For the second evaluation (comparison to similar system), participants are given three types of smartphone based hearing aids: HearMe (the system we build), i-Hear, and Petralex. There are four categories we are interested in: sound quality, system flexibility, ease of use, and overall system. All of the participants will try all three applications, and we asked them to name the best applications for each categories. The result of the comparison evaluation can be seen in Table 4.

Table 4: Comparison Experiment Result

Category	HearMe	i-Hear	Petralex
Sound Quality	5	1	1
System Flexibility	7	0	0
Ease of Use	2	2	3
Overall System	5	0	2

From the comparison results, system built (HearMe) excels in sound quality, system flexibility, and best application overall. Most participants recognize HearMe sound quality is better as there are less noise than other application. In this case, noise gate module helps to greatly reduce the noise. System built even have less noise than hearing aids participants currently use. HearMe also very flexible as each user can set their own gain for each frequency range. Other application only have one gain settings which will be applied to signal in every frequency. This type of application cannot cover every hearing impaired characteritics. For ease of use category, Petralex is more preferable to participants as Petralex provide hearing tests when the application starts. The gain settings then will be set according to the hearing test results. Hearing impaired won't have to change the settins manually. Some participants also prefer manual setting as the hearing tests are a bit complicated, takes time, and the result is not very accurate to their hearing loss.

5. CONCLUSION

This paper proposes a novel method to build hearing aids based on smartphone. The proposed approach mimics how humans ear works to help people with hearing loss. The result of the evaluation are:

- Hearing impaired are able to listen to surrounding sound using the help of system built. This is proven by the satisfying functionality test accuracy average of 79.3%.
- Environment doesn't contribute significantly to hearing impaired ability to listen using the smartphone based hearing aids. This is proven by the small accuracy difference between quiet and crowded environment on average: 4.29%.
- Hearing impaired can tune the gain parameter themselves on each channel that suits best with their hearing loss characteristics. This settings make the system excels in flexibility as all participants agree that HearMe is more flexible than similar system.
- Application doesn't need internet connection to process the sound. This is very helpful to people living in an area where internet is still rare
- Noise gate is doing a great job on blocking noise compared to hearing aids that have constant buzzing sound which hearing impaired doesn't want to hear. This results in better sound quality compared to similar system
- System built is more preferable than similar hearing aids as 5 out of 7 participants prefer HearMe on overall system

In the future, there are some aspects that can be improved from algorithm and practical points of view. The previous apps cannot modify sound from phone call as the apps automatically closed when there's phone call. We would like to make the apps able to modify phone call, so hearing impaired can communicate using phone calls. From the evaluation, we also know that by using this apps, the user doesn't know where does the sound come from. In hearing aids, user can know the direction of the sound as the sound emitted is stereo, while in the apps the sound produced is mono. Currently the apps only works when microphone is inserted to the smartphone. If microphone isn't inserted, the apps will produce a buzzing sound due to the output

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sound will be received again by the smartphone as input sound. We would like to apply feedback cancellation algorithm to prevent this buzzing sound from happening.

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