

Design And Development of Improved Digital Hearing Aid

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ABSTRACT: Hearing Aid is a small amplifying device which fits on the ear, worn by a partially deaf person. In this paper, an adaptive hearing aid device was designed and developed to improve the hearing of the deaf, which should help correct the speech deficiency and by intelligently increasing or decreasing the loudness of the acoustic signal. An Electret microphone was used for capturing sounds from all directions. The signal from it was pre-amplified using a low pass filter. Analog to Digital Converter (ADC) and Digital to Analog Converter (DAC) Circuit was developed for the conversion of signals to electrical and back to analog. A Dynamic Range Compression (DRC) Algorithm was developed to map the wide dynamic range of input signal into the reduced dynamic range of hearing for the impaired person. An Adaptive Feedback Cancellation (AFC) algorithm was developed to overcome the acoustic feedback problem in digital hearing aid. The developed codes were encoded into an integrated circuit. The software was simulated and configured with the target hardware. The developed digital hearing aid was measured using microphone coupler. The developed digital hearing aid was fabricated into the mould, miniaturized and packaged. It was tested with a hearing impairment patient and the patient was satisfied with the device

KEYWORDS: Digital Hearing Aid, Microphone Coupler, DRC Algorithm, AFC Algorithm, ADC, DAC

Date Of Submission:08-10-2018

Date Of Acceptance: 23-10-2018

I. INTRODUCTION

The human body is built in such a way that the organs compliment themselves to achieve common goals whether they are internal or externally fitted. The ones we see externally also have internal connections that make the human body function as a network. The absence or malfunctioning of any renders the operations of the body inefficient. While the malfunction of some may not be very noticeable, the eye, ear and throat are so vital that any fault in them is easily observable. The blind person needs a guide, the deaf while seeing does not comprehend or communicate with the environment freely and easily. The blind may not venture walking alone, however the deaf may but at a very high risk as only the danger seen can be avoided. Although, many events are noticed due to the sound made reaching to them. For instance, a vehicle that fails break can attract shout from passerby which when heard, could help them to react but the deaf will not and even when heard may not know the direction to move to avoid accident.

There is absolute need for those living with hearing disabilities and already using some form of hearing aids to improve their hearing abilities so that they can walk freely in both streets and major roads without endangering their lives, especially getting involved in accidents [1]. Apart from that, hearing abilities of infants born with such disabilities also need to be improved, so that they can learn how to talk because, it is what they hear that prompts their response. There are conventional hearing aids that are already in existence but, have some shortcomings in terms of function and capacity. Although some have multiple channels which help in different frequency range, which can be adjusted with potentials but they are not electronically programmable. Also, the existing systems are powered by batteries that dissipate within short intervals. One can ask the question; for how long does the battery stay alive? So power consumption is one of the drawbacks of these conventional hearing aids / devices.

One can remedy the effect of high power consumption by opting for digital hearing aids but what about the ability of the device to filter off noise such as loud automobile loud buzz, so that the user or patient would be able to operate in an open and noisy environment. It is understandable through observation that transmission

range and noise plays a major role in determining the effectiveness of any form of hearing aid and the extent to which a patient hearing ability is improved. Hence, the need for a hearing aid that is adaptive which can intelligently increase or decrease the loudness of the acoustic signal within any type of environment, and at the same time adjusts the transmission range with respect to the source of the sound.

II. LITERATURE REVIEW

2.1 The Hearing Organ

In the human body system, ear is the organ for hearing. The main function is transmitting and transducing sound to the brain through its different parts. The ear can anatomically be divided into three parts; (fig1)

- a. The outer ear
- b. The middle ear
- c. The inner ear

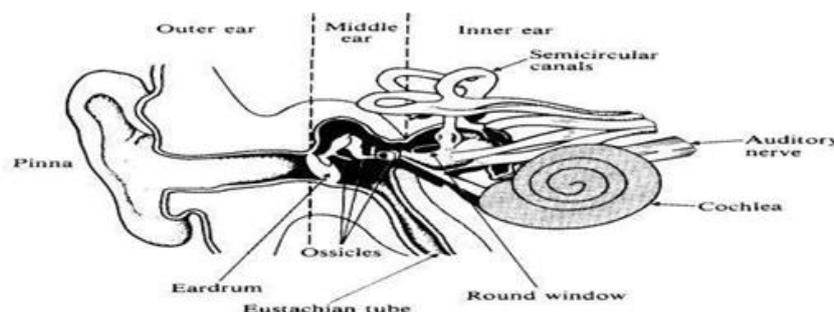


Fig 1 Ear structure

The outer ear which is the input transducer consists of the pinna, the external auditory canal and the tympanic membrane (eardrum). Here, the pinna picks up the sound waves. The waves pass through the auditory canal and causes the eardrum to vibrate. In the middle ear, the sound wave that made the ear drum to vibrate in turn causes the ossicles, the anvil and the stirrup; which are known as the small bones to move[2].

Also in the inner ear, the vibration moves through the oval window, through the fluid in the cochlea; which is one of the constituents of the inner ear causing the stimulation of thousands of tiny hairs. The stimulation of the hairs apparently results in the transformation of those vibrations into electrical pulses that the brain receives.

2.1 Causes of Deafness

Deafness basically is loss of auditory functions depending on the severity of the hypoacusis [3]. Deafness can be categorized in levels of mild, moderate or total deafness.

Deafness can be classified under

- a. Conductive deafness
- b. Sensor neural deafness and
- c. Mixed deafness

Congenital atresia, tumors, stenosis etc. of the ear canal when defective causes the sound conducting mechanism of the ear to be abnormal, leading to conductive deafness. In the middle ear, acute and chronic otitis media, serous and suppurative otitis media can cause abnormality in the cochlea, auditory nerve, neural pathway etc. leading to sensorineural deafness[4].

Combination of both conductive and sensorineural abnormality leads to mixed deafness. There are many other abnormalities that can lead to the above mentioned classification of deafness that are not mentioned here.

2.2 Present Curative Measures

1. Hearing Aid

Presently the most effective and efficient way of handling deafness is by the use of hearing aid devices. The hearing aid device is basically what brings in sound to the ear more effectively. The basic function of a hearing aid is to amplify and couple sound wave to the ear[5]. The basic components that make up the system are:

- i. The microphone
- ii. Amplifier and
- iii. Receiver.

Generally the sound wave, which can also be referred as the acoustic signal is converted to electric signal. That electric signal is amplified and reconverted to acoustic energy. The microphone is the input transducer that picks up the acoustic signal and converts it to electrical signal. The Amplifier is the signal processor that does the amplification of the electrical signal. At the end, the receiver which is the output transducer, usually the earphone reconverts the amplified electrical signal into acoustic energy as before.

The hearing aid device is so miniaturized in such a way that it can be body worn, can be behind the ear and can be in the ear canal. These correspond to three basic types of hearing aid. Others include: contralateral routing of sound hearing aid, telephone hearing aids and group hearing aid for auditory framing and for educational institutions.

2. Cochlear Implant

Some deaf persons do not respond to amplified sound stimuli especially in patients with bilateral severe sensorineural abnormality. Just like the hearing aid, the cochlear implant consists of stimulating auditory system as against surviving hair cells used by hearing aids [6]. Stimulation of electrical signal is important here and easy because of the nature of the cochlea. The cochlea behaves as an output transducer that changes the mechanical energy of sound vibration into electrical impulses. Obviously these impulses are the signals which are transmitted along the nerve.

The way the implant works is not complex but very simple as follows:

- i. The acoustic signal is picked up by the microphone
- ii. The signal is converted to electrical voltage and sent along the cable to the speech processor
- iii. The speech processor manipulates the signal according to coding strategies develop to induce optimal hearing with the cochlear implant.
- iv. The transmitter receives the coded signal via the cable
- v. It is now the duty of the transmitter to transfer these signals in accordance with the required energy by the implanted electronic via the intact skin to the implanted receiver.
- vi. Those implanted receiver and stimulator decode the signal and sends a logic pattern of small electrical impulses to the electrodes in the cochlea
- vii. Those small pulses conducted by the electrode contacts, stimulate the spinal ganglia at various sites and different parts of the auditory nerves are stimulated in accordance with the pitch of the sound.
- viii. Finally, the brain receives the nerve impulses and does the interpretation. The brain interprets them as sound which the patient hears

III. METHODOLOGY

The design work was achieved in two parts

1. Hardware design
2. Software design

The block diagram of the designed system is as shown in fig 2.

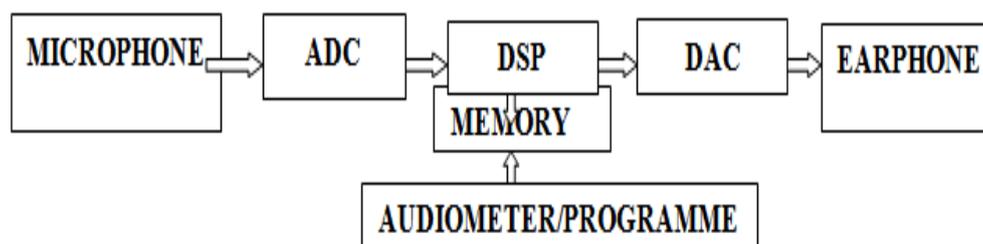


Fig 2. System Block Diagram

Electret microphone which is a type of electrostatic capacitor based microphone for capturing sounds from all directions was used.

The Electret microphone has 20-20KHz frequency response which is the highest frequency most humans can hear at -44dB sensitivity. The signal from it was preamplifier using NPN Bipolar junction transistor. The signal was then low pass filtered to 20KHz to avoid aliasing during sampling before it enters the Analog to Digital Converter (ADC). The ADC transformed the signal into electrical signal. The digital signal display is the heart of the hearing aid because it is where the sound is amplified to a certain frequency band of interest without causing

distortion, compresses the loud signal but makes the sounds become clearer and speech easier to hear and understand.

A Dynamic Range Compression (DRC) Algorithm in fig 3 was used in the DSP to map the wide dynamic range of input signal into the reduced dynamic range of hearing impaired person to enhance the speech signal to be above the hearing threshold but at the same time below the discomfort level.

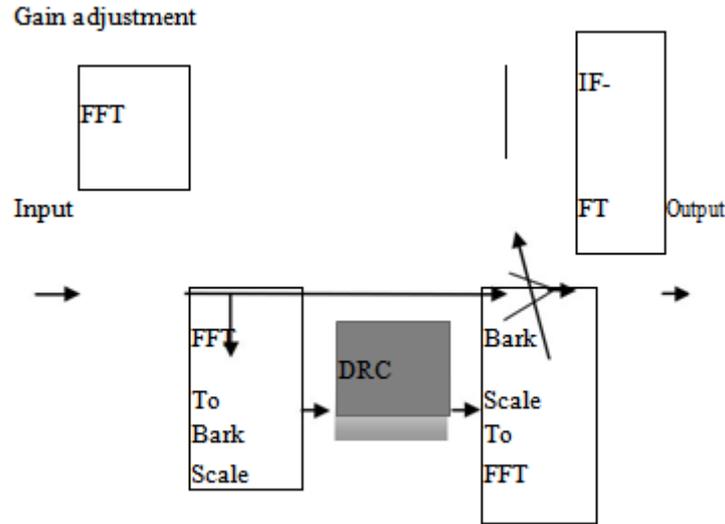


Fig 3 DRC algorithm

Parameters considered in developing the DRC Algorithms are Compression Threshold (CT), Compression Ratio (CR), Attack Time (AT), Release Time (RT) and DRC Gain (G). CT is defined in dB and determines the point where DRC becomes active. Compression Ratio (CR) determines the degree of compression required for input audio signal. Attack Time (AT) is the time taken by compressor to increase the level of input audio signal. Release Time (RT) is the time taken by the compressor to decrease the level of input audio signal. DRC Gain (G) is the speech DRC gain selected by the user.

The DRC input signal is given by

$$DRC(k,l)ip = 20\log_{10} \frac{DRC(k,l)}{P_{ref}} \dots\dots\dots(1)$$

k is used to indicate that linear frequency is now mapped to Bark Scale and P_{ref} is the reference sound pressure

The DRC output signal is given by

$$DRC(k,l)op = \{p_{lin}(k,l), \text{ if } DRC(k,l)ip < CT \text{ or } p_{cp}(k,l), \text{ if } DRC(k,l)ip > CT \} \dots\dots(2)$$

Where

$$p_{lin}(k,l) = DRC(k,l)ip + (G) \dots\dots\dots(3)$$

$$p_{cp}(k,l) = CT + \frac{1}{CR}(DRC(k,l)ip - CT) + (G) \dots\dots\dots(4)$$

By this process, volume of loud sounds are reduced while the quiet sounds are amplified. The amplified digital signal is reconverted into acoustic signal by the digital to analogue converter (DAC)

u(t)

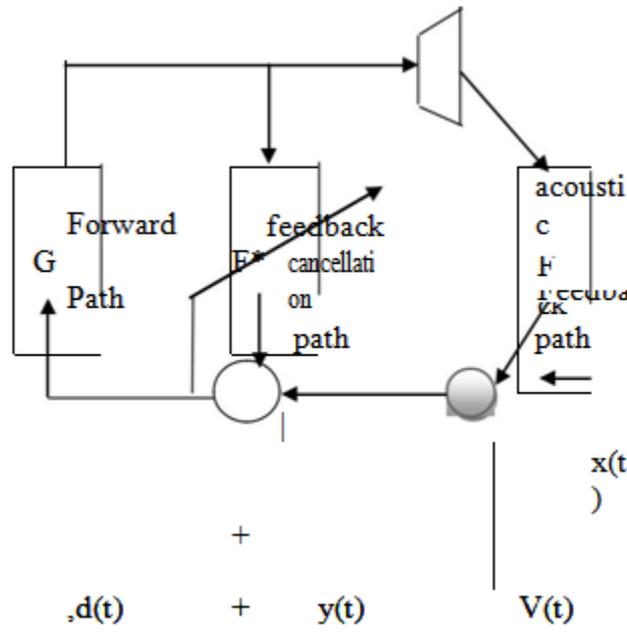


Fig. 4: Adaptive Feedback Cancellation (AFC) algorithm

Thereafter an Adaptive Feedback Cancellation (AFC) algorithm in fig 4 was developed to overcome the acoustic feedback problem in the digital hearing aid. In this algorithm an adaptive finite impulse response filter (FIR) was used to model the acoustic feedback path which predicts the feedback signal in the microphone signal. The predicted feedback signal is then subtracted from the microphone signal. The microphone signal y(t) is given as

$$y(t) = v(t) + x(t) = v(t) + F(h,t) u(t) \dots \dots \dots (5)$$

where x(t) is the feedback signal, v(t) is the near end signal, u(t) is the loud speaker signal, h is the time shift operator, t is the discrete time variable and F(h,t) is the feedback path between the loudspeaker and the microphone.

The forward path (G) maps the microphone signal to the loudspeaker signal while the feedback canceller (F*) produces an estimate of feedback signal that is further subtracted from microphone signal.

Hence the feedback compensated signal d(t) is given by

$$d(t) = v(t) + F(h,t)u(t) - F^*(h,t)u(t) \dots \dots \dots (6)$$

Finally, the output transducer which is the earphone transmits the acoustic signal to the ear.

IV. RESULTS

The Maximum Output Sound Pressure Level(Max OSPL), High Frequency Average Output Sound Pressure Level (HFA OSPL), Full On Gain (FOG), SPL for an Inductive Telephone Simulator(SPLITS), Simulated Telephone Sensitivity(STS),Frequency cut-off, Total Harmonic Distortion(THD) of the developed digital hearing aid were measured using microphone coupler. The recorded results from the measurement are in table 1

Table 1 Digital Hearing Aid Electroacoustics Measurement

Parameters	Measured value
Max-OSPL90	122dB
HFA-OSPL90	109dB
HFA-FOG 50dB	52dB
HFA-SPLITS	107dB
STS	4dB
High Frequency Cut Off	7.2kHz
Low Frequency Cut Off	300Hz
THD at 70dB	2.1%
Current drain	12mA
Attack time	1.2ms
Release time	380.5ms

Input Noise	38dB
Frequency Response	390-4500Hz

It conformed to the standard of International Electrotechnical Commission (IEC) of hearing aid electroacoustics measurement. The developed digital hearing aid shown in fig 5 was tested and certified with someone that has hearing impairment. She was able to hear all the conversation even in the noisy environment.



Fig 5 Developed digital hearing aid

V. CONCLUSION

A handy digital hearing aid was designed, developed and tested. With the adaptive algorithms used, it gave better hearing improvements with high value of signal to noise ratio. It proved to be robust and relatively cheaper than the one in the market. Future design should be based on wireless digital hearing aid

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Eke, " Design And Development of Improved Digital Hearing Aid" American Journal of Engineering Research (AJER), vol. 7, no. 10, 2018, pp. 249-254