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DIGITAL HEARING AID –A REVIEW

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Abstract: In the rural sector of India there is a large number of hearing impaired persons. With moderate or acute hearing loss Hearing impaired persons, suffer from opportunities for communicating with others Presently the conventional analog hearing aids have a number of functional limitations such as fixed frequency, lack of flexibility, can’t distinguish speech from noise, etc. The Digital hearing aid can solve these problems. This device is fully programmable having features like Wide Dynamic Range Compression, Feedback Suppression, and Adaptive Noise Cancellation.

Keyword: Digital signal processing, noise suppression, dynamic range compression.

I. INTRODUCTION

The recent studies shows that nearly every fourth person suffers from hearing problems. For this reason the necessity of the hearing aid is come forward. Some common problems can be solved with the help of conventional analog hearing aid .But some are very acute and it require the programmable hearing device.

II. CAUSES OF HEARING DISABILITIES

There are different types of hearing impairment due to different causes. The commonly occurring problem is due to the ageing effect.With the types of the different hearing losses the different types of losses are present.

There are following types of hearing losses.

1.*Conductive hearing loss*

Causes :

- Some Middle ear infections
- Sevier Head injury
- Birth defects
-

2. *Sensorineural hearing loss*

Causes :

- Aging effect
- Heredity/genetics
- Head injury

3. *Neural hearing loss*

Causes :

- Heredity / genetics
- Sevier Head injury
- Tumor

4. *Mixed hearing loss*

Causes:

- Combination of conductive and sensory causes

III. TYPES OF HEARING AID

There are different types of hearing aid which are given bellow.

1. CIC - Completely In The Canal
2. ITC - In The Canal
3. ITE - In The Ear
4. BTE - Behind The Ear

Problems faced by people with hearing impairment-

1. Decreased Audibility- The decreased audibility can be formed due to the increased Hearing Threshold. Hearing threshold means that frequency at which the person can hear without using hearing aid. Sometimes the person can hear the sound but cannot differentiate in it. They found the sound is too loud or not clear enough.

2. Decreased Dynamic Range- The sound level difference between the ULL and the hearing threshold is called the dynamic range Uncomfortable Loudness Level (ULL) is the maximum sound level the patient can hear comfortably at that frequency. Normally the dynamic range will be less than that of a normal ear. When the dynamic range is less, then the amplification required to make weak sounds audible. But this will cause medium sounds uncomfortably loud.

3. Decreased Frequency Resolution- In some types of losses it is observed that a person has decreased frequency resolution. This can be caused by damaged cochlea .Even for normal ears the frequency resolution will be less at higher sound levels. If the frequency resolution is

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sufficiently less, relatively intense low frequency parts of speech may mask weaker high frequency components.

4. Decreased temporal Resolution- Sometimes the intense sounds can mask weaker sounds that immediately follow them. This type of masking is present for people with sensorineural hearing loss.

Limitations of Conventional Analog Hearing Aids

The convensional anolog hearing aid have the following drawbacks:

- The anolog hearing aid provide the same amplification for all the frequencies present in the spectrum But the hearing loss may be different for different frequencies. So the aid should consists the variable gain.
- With the help of anolog hearing aid we can not program the aid which characteristics to suit a particular user's hearing profile.
- Due to the amplification of all audio signals of frequency and it will increases the sound levels of high intensity signals generates high values resulting in uneasiness for the user. In digital hearing aid the gain can be adjusted accordingly.
- In the presence of background noise the user is unable to understand the speech signals in it.

IV. HARDWARE DESIGN OF THE SYSTEM

Here in the circuit the speech signal is captured by microphone and is converted to electrical signal. The magnetic field produced by hearing aid is converted into electrical signal. It can be selected when there is lot of background noise. The signal is amplified by the pre-amplifier. An ADC is required to convert the analog signals to digital signals. Further processing is done in the digital domain.
By using the ADC the analog signal converted into digital signal and fed to the FPGA block. Where it will processed and sampled. The amplification or compression of the signal is done in the FPAG programming block. Then the processed signal is again converted into the analog form with the help of the DAC.

The Block diagram of the entire circuit is as shown in fig.

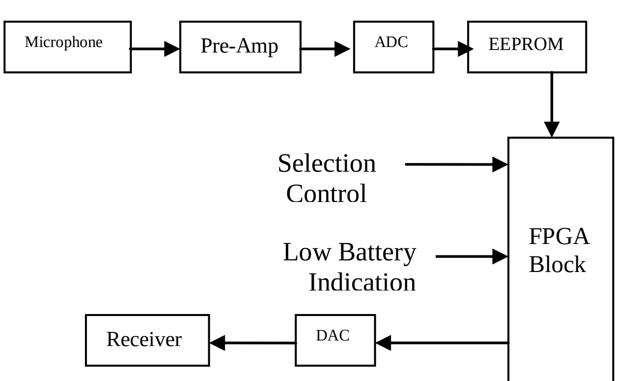

Fig.1 : Block Diagram

With the help of FPGA block various functions can be carried out. The DSP uses the data stored in EEPROM to implement the DSP Algorithm and process the digital signal. There are basically two approaches in DSP. One is the Time Domain approach and the other Frequency Domain approach. We follow the Frequency Domain approach in the development of DSP algorithm for hearing aid. The output of ADC is read and stored in memory. Once the input buffer is full, it is read, processed and the output samples are generated. The processing should be completed before the input buffer is again filled.

V. SOFTWARE DESIGN OF THE SYSTEM

Altera DE2 board and Xilinx board becomes one of the most widely used FPGA board for development of FPGA design and implementation. The Purpose of this board is to provide the ideal path for learning about FPGA, digital logic and computer organization. The board offers a large set of features that make it suitable for use in laboratory environment for the variety of design projects and for the development of digital systems. Altera provides various supporting material for DE2 board, including demonstrations, laboratory exercises and tutorials.

VI. DESIGN FLOW OF THE SYSTEM

The flow of the project is in three phases:
1. **Dynamic Range Compression**
2. **Noise Suppression**
3. **Feedback Reduction**

It includes:-

1. Dynamic Range Compression: Uncomfortable loudness level(ULL) at a particular frequency is the maximum sound level the patient can hear comfortably at that frequency.
The difference between the ULL and hearing threshold is called as **Dynamic Range.**
There are 3 types of Dynamic range compression
A. Low level Compression-

- Amplify factor is reduced for signal level below the compression threshold.
- Linear amplification is provided for signals above the compression threshold.
- Leads to squashing dynamic range for weak to moderate signals.

B. High level compression-

- Amplify factor is reduced for signal level above the compression threshold.
- Linear amplification is provided for signals below the compression threshold.
- Leads to squashing dynamic range for moderate to intense sound.

C. Wide Dynamic range compression

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- Compression is applied over a wide range of input sound level.
- There are no sound levels for which the o/p levels need to be squashed together closely.

compression reduces the volume of loud sounds or amplifies quiet sounds by narrowing or "compressing" an audio signal 's dynamic range.

2. Noise Suppression

There is a one more drawback is that the present day hearing aids cannot distinguish between speech signals and noise. So both noise and speech signals are amplified at the same time. As a result the signal-to- noise ratio is improved. Due to this reason for the speech processing, the input signal is separated into different frequency channels, each channel is analysed individually. Speech signal has has some structural difference from most noises. With the help of Digital Signal Processing we can identify this structural difference ,which is not possible with analog technology. So for that we have to decide whether the signal is speech or noise for each frequency range. From that signal each frequency range is then amplified or attenuated accordingly. With the help of this processing technique it is possible to suppress or remove background noise without affecting the speech signals. It is called as Adaptive Noise Cancellation.

3. Feedback Reduction

Acoustic feedback is another common problem existing with present day hearing aids. In the design of hearing aid the receiver and transmitter are are place on a small distance from each other so some of the output of hearing aid may get back to the input of the aid. The signal feedback will be processed along with the incoming sound.

One common method to avoid feedback oscillation is to adjust gain frequency response. In single channel hearing aids this is achieved by reducing the overall gain. Also we can design the multichannel hearing aids in which it is possible to reduce the gain only at those frequencies where the feedback occurs. It is also possible to limit the maximum gain in each channel.

VII. CONCLUSION

We implemented an FPGA-based Digital Hearing Aid which employs a multiband compression system algorithm.Which proposed that compression is applied separately in two or more frequency bands. The design was carried out using Verilog VHDL and the implementation was based on the FPGA development board. We studied and profiled the speech processing algorithm implemented in the software version and designed a highly-parallel architecture. The digital hearing aid is useful to overcome all the limitations of anolog hearing aid.

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BIOGRAPHY

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