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FPGA Based Signal Processing Implementation for Hearing Impairment

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Abstract: *Hearing aids partially overcome auditory deficits and normally employed for the hearing impaired people to compensate hearing loss. This paper takes an approach to design a system which have a digital filter bank which can separate the input speech signal into different bands and automatically gain is adjust for individual band. The system is developing on Field Programmable Gate Array (FPGA) platform. The system consists of analog-to-digital (A/D) converter, digital filter bank, gain control and digital-to-analog (D/A) converter. The A/D converter converts the analog speech signal into a digital speech signal. After that input signals gets process by digital filter bank and gain control. For a person with hearing impairment the range of levels between weakest sound that can be heard and the most intense sound that can be tolerated is less than normal listener. For compensating this, hearing aids amplify weak sounds more than they amplify intense sound. The D/A converter converts the output processed signal into analog speech output signal. The speaker converts the analog speech signal into an acoustical output signal that is directed into ear canal of the hearing instrument user.*

Keywords: *FPGA platform, Digital filter bank, Gain Control, A/D and D/A converter.*

1. Introduction

Normally human beings received information in form of auditory, visual and sensory responses. Human can communicate by taking complicated perceptual information from the outside world through hearing sense and then that information is interpreted in a meaningful way. Auditory system is one of them. Mainly the human auditory system consists of outer ear, middle ear and inner ear. The outer ear is responsible for gathering sound energy and funnel it to ear drum, the middle ear which acts as a mechanical transformer and the inner ear where the auditory receptors (hair cells) are located which accepts the mechanical signal and convert it into electrical signal transfers it to the brain. So, for person suffering with severe sensory-neural deafness, the auditory nerve fibers using electrical stimulation are design [1]. Over the years several hearing aids were developed which have following features in common: a microphone that picks up the sound and a signal processor that processes the audio signal & converts it into the audible form for hearing impaired person. Here, the speech processor is responsible for decomposing the input audio signal into different frequency bands [2]. The designers of hearing aids are faced with challenge of developing signal processing strategies that can extract sufficient amount of spectral information from the speech signal. There are different types of difficulties occurs during signal processing such as: The compression algorithm is a system dependent characteristics since the core of used hearing aid forces the set of allowed algorithms. Apart from compression, the main parameters to program are the Noise Suppression techniques and Feedback Reduction algorithms [7]. Noise Reduction is an important stage in the hearing aid signal processing since hearing-impaired people have to understand speech with background noise. The problem of Feedback Reduction produced when the sound goes from the loudspeaker to the microphone. It always limits the maximum gain and reduces the sound quality. So, these factors demand to develop high performance speech processor for hearing aids.

2. Basics of Digital Hearing Aid

All modern hearing aids have the following architecture and functional blocks. There are different strategies such as Continues Interleaved Scheme (CIS), *n*-of-*m*, spectral peak (SPEAK), advanced combination encoder (ACE) and Hi-Resolution (HiRes) [1]. The *n*-of-*m*, SPEAK and ACE strategies each use a channel-selection scheme. In these techniques for the different channels envelope signals are scanned prior to each frame of stimulation across the intra-cochlear electrodes.

But amongst all these, the CIS strategy is best. With present-day hearing aids the Figure 1 shows one of the simpler and most effective approaches for representing speech and other sounds. Normally this strategy is used as the default strategy [1]. A bank of bandpass filters is used to filter out the speech or other input sounds into bands of frequencies from CIS strategy. Then at corresponding electrodes in the cochlea different bands shows different envelope variations.

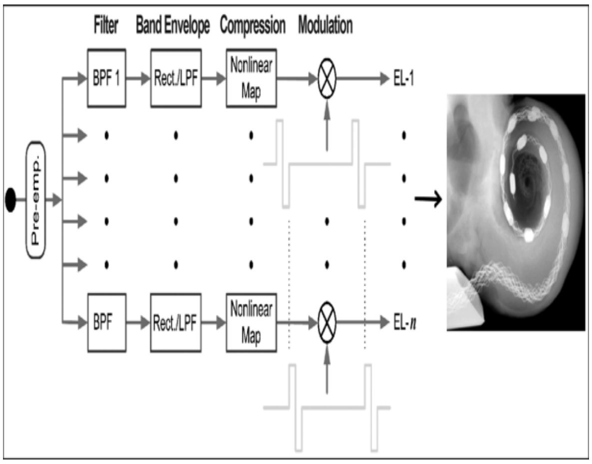

Figure 1. *Block diagram and signal processing in the continuous interleaved sampling (CIS) strategy.*

Prior to the modulation, a nonlinear mapping function is used to extract the envelope signals from the bandpass filters which get compressed with in order to map the wide dynamic range of sound in the environment (up to about 100 dB) into the narrow dynamic range. For the different channels and corresponding electrodes the pulse trains are interleaved in time so that the pulses across channels and electrodes are non-simultaneous. The corner or “cut-off” frequency of the low-pass filter is typically set at 200 Hz or higher in each envelope detector, so that the fundamental frequencies (F0) of speech sounds, e.g., 120 Hz for male voices, are represented (exclusively) in the modulation waveforms. The acronym for CIS is continuous interleaved sampling strategy of the (compressed) envelope signals. Till date CIS scheme uses between 4 to 24 channels for signal processing the speech or other sound signals [8-9]. (CIS processors are often described as having a small number of channels and associated sites of stimulation, e.g., six to eight, but this is incorrect. Itself the strategy does not place a limitation on the number of channels; Up to date CIS implementation have used as many as 24 channels.)

3. Signal Processing Techniques

Various types of difficulties occurs during the signal processing in hearing aids. For avoiding those difficulties following techniques are used.

3.1 Dynamic Range Compression

At a particular frequency the maximum sound level that can be hear by patient comfortably is called the Uncomfortable loudness level (ULL).

So, the difference between the ULL and hearing threshold is called as Dynamic Range [7].

There are 3 types of Dynamic range compression techniques:

1. *Low level compression*
   - For signal level amplify factor is reduced below the compression threshold.
   - For signals above the compression threshold linear amplification is provided.
   - This compression leads for weak to moderate signals.
2. *High level compression*
   - For signal level amplify factor is reduced below the compression threshold.
   - For signals above the compression threshold linear amplification is provided.
   - This compression leads for moderate to intense signals.
3. *Wide dynamic range compression*
   - Compression is applied over a wide range of input sound level.
   - There are no sound levels for which the output 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.

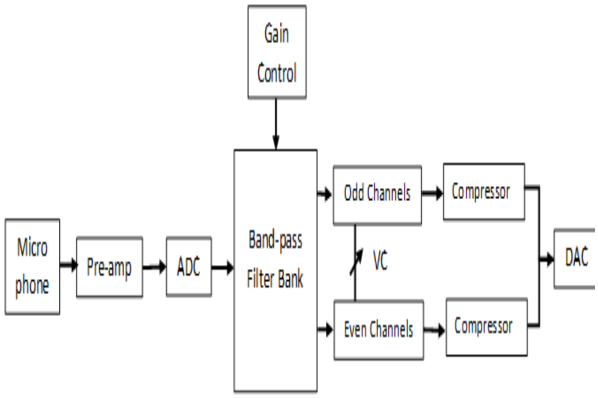

Figure 2. *Block diagram of signal processing algorithm*

3.2 Noise Suppression

There is a major drawback of present day hearing aids is that they cannot distinguish between speech signals and noise. So the amplifier treats both noise and speech signals in the same manner, and thus it amplifies both speech and noise, As a result no improvement in the signal-to- noise ratio is done. Speech signal cannot be clearly perceived in many cases. For speech sensitive processing, after separating the input signal into different frequency channels, each channel is analysed individually. Speech has a temporal structure different from most noises. With the help of Digital Signal Processing we can identify this different temporal structure to a degree, which is not possible with analog technology. In each of the frequency ranges these results are used to decide whether the signal is more likely speech or noise. From that signal each frequency range is then amplified or attenuated accordingly. This processing technique makes it possible to suppress background noise without affecting the speech signals [7].

3.3 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 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. In multichannel hearing aids 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 [7].

4. Hardware Design of the System

The sound is captured by microphone and is converted to electrical signal. Tele-coil senses magnetic field produced by hearing aid converts it into electrical signal. It is used as auxiliary input and is selected when there is lot of background noise. The signal is amplified by the pre-amplifier [7]. An Analog to Digital Converter is required to convert the analog signals to digital signals. Further processing is done in the digital domain. By using over-sampling technique along with noise-shaping circuit, sigma delta ADC increases the effective resolution. The Sigma-Delta ADC converts the signal into digital signals. The FPGA block reads the data and stores it. It manipulates the digital signal and sends it to the Class D amplified receiver, which converts the electrical signal to sound. HI-PRO contains level converter and the necessary logic for enabling serial communication between PC and Hearing Aid [1].

The various functions carried out by the FPGA block are discussed below. Here, Band-pass filter bank is used to separate the input signal into different frequency bands. The gain control block adjust the gain for each band. The odd channel formed by the summation of odd channel filter bands and even channel by even channel filter bands. The gain in odd and even channels is control by volume control (VC). Compressor block compresses the output audio signal. Then final output signal is formed by summation of output of odd and even channels. Further the digital signal gets convert into analog form by using DAC [1, 7].

5. 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.

6. Results

The microphone takes the input from the environment which is an analog form. Then A/D converter converts the analog signal into digital form. Then digital filter bank separate input signal into multiple channels each having particular band of frequencies. After that gain is adjusted for required band automatically. Then signal is get compressed and finally D/A converter convert the processed signal into digital form to anaog form. The figure 3 shows the output processed signal in matlab software.

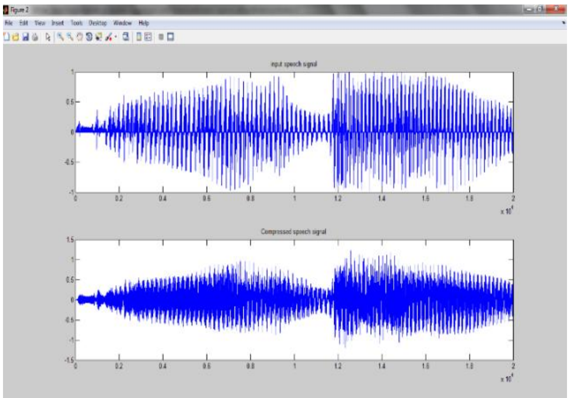

Figure 3. *Final output signal on matlab software*

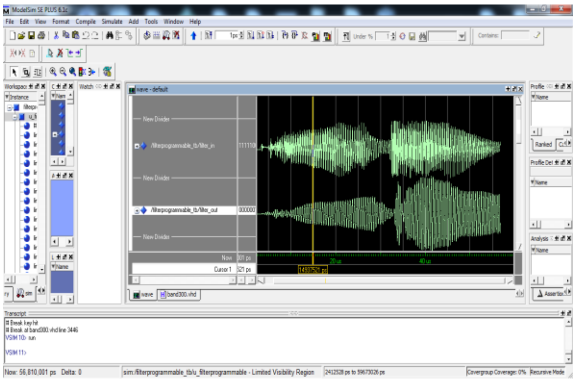

Figure 4. *Final output signal on modelsim software*

After calculating the values of coefficients for digital filter bank from matlab software. The same system is develop on FPGA platform. The Xilinx software is used to develop the system and its interactive software modelsim is used to get final output. The figure 4 shows the final output of system based on FPGA.

7. Conclusion

For a person with hearing impairment, the hearing aids are the only device which make them audible to the outside environment. The FPGA-based Digital Hearing Aid system which employs compression system algorithm. This proposed that compression is applied separately in multiple frequency bands. The design was carried out using VHDL or MATLAB the implementation was based on the FPGA development board. This paper presents systematic and comprehensive design and specifications of digital hearing aid. It is fair to conclude that the digital hearing aids not only has a long and distinguished history but also a bright future as it continues to broaden its utility for treatment of a wide range of hearing impairment and to serve as a model to guide development of other neural prostheses.

References

[1] Fan-Gang Zeng, Stephen Rebscher, William Harrison, Xiaoan Sun and Haihong Feng, “Cochlear Implants: System Design, Integration, and Evaluation”, in proc. IEEE Reviews in Biomedical Engineering, vol. 1, pp. 115 – 142, 2008.

[2] P. Mahalakshmi, M. R. Reddy, “Signal Analysis by Using FIR Filter Banks in Cochlear Implant Prostheses”, in proc. International Conference on Systems in Medicine and Biology, IEEE, pp. 253 - 258, December 16-18, 2010.

[3] Ruan A.W., Liao Y. B., Li P. , Li J. X., “An ALU-Based Universal Architecture for FIR Filters”, in proc. International Conference on Communications, Circuits and Systems, IEEE, pp. 1070 – 1073, 2009.

[4] Nekoei F., Kavian Y. S., Strobel O., “Some Schemes Of Realization Digital FIR Filter On FPGA For Communication Application”, in proc. 20th International Crimean Conference on Microwave and Telecommunication Technology, pp. 616 – 619, 2010.

[5] Peng Qiao, Henk Corporaal, Menno Lindwer, “A 0.964mW Digital Hearing Aid System”, in proc. Design, Automation and Test Europe Conference and Exhibition (DATE), pp. 1 – 4, March 14–15, 2011.

[6] D. Tharini, J. Prathishkumar, “21 Band 1/3-Octave Filter Bank for Digital Hearing Aids”, in proc. The International Conference on Pattern Recognition, informatics and Medical Engineering, pp. 353 – 358, March 21-23, 2012.

[7] Lunner T., Hellegren J., Arlinger S., Elberling C., “Non-Linear Signal Processing in Digital Hearing Aids” in proc. Scand Audiol IEEE , pp. 40 – 49, 1998.

[8] X. Jiang, Y. Bao., “An Design of the 16-order FIR Digital Filter Based on FPGA” in proc. International Conference on Information Science and Engineering (ICISE), IEEE, pp. 489-492, 2012.

[9] N. S. Lawand, W. Ngamkham, G. Nazarian, P. J. French, W. A. Serdijn, G. N. Gaydadjiev, J. J. Briaire and J. H. M. Frijns. “An improved system approach towards

future cochlear implants", IEEE Conference on EMBS, vol. 13, pp. 5163-5166, July 2013.

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