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Reference documents

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79 Sentences were found in a text with the title: „***ACTA-2020-2-17.pdf***", located at:
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http://dspace.uah.es/dspace/bitstream/handle/10017/20133/David Ayllon Thesis.pdf;jsessionid=C2D9013D8CBA4F558D44241457E71FF3?sequence=1

2 Sentences were found in a text with the title: „***01_M004_46589.doc***", located at:
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2 Sentences were found in a text with the title: „***Multi-UAV Deployment for Coverage with Spatio-Temporal Specifications***", located at:
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http://citeseerx.ist.psu.edu/viewdoc/download?doi=10.1.1.56.6260&rep=rep1&type=pdf

2 Sentences were found in a text with the title: „***209617.pdf - 004635265a022a281a000000.pdf***", located at:
https://www.researchgate.net/profile/Jeng-Shiou_Leu/publication/220243275_Crafting_a_Real-Time_Information_Aggregator_for_Mobile_Messaging/links/004635265a022a281a000000.pdf

2 Sentences were found in a text with the title: „***209617.pdf***", located at:
http://downloads.hindawi.com/journals/jcnc/2010/209617.pdf

2 Sentences were found in a text with the title: „***54ceb6e90cf298d656620392pdf - 54ceb6e90cf298d656620392.pdf***", located at:
https://www.researchgate.net/profile/Francois_Rigaut/publication/260961563_Estimation_of_the_adaptive_optics_long-exposure_point-spread_function_using_control_loop_data/links/54ceb6e90cf298d656620392.pdf

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http://citeseerx.ist.psu.edu/viewdoc/download?doi=10.1.1.189.1834&rep=rep1&type=pdf

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https://www.fda.gov/medical-devices/hearing-aids/types-hearing-aids

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2 Sentences were found in a text with the title: „***Types Of Hearing Aids | Alternatives For Seniors***", located at:
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https://www.rxlist.com/script/main/art.asp?articlekey=114512

2 Sentences were found in a text with the title: „***Hearing Aid – Helping People Communicate***", located at:
http://helpingpeoplecommunicate.com/hearing-aid/

2 Sentences were found in a text with the title: „***Hearing Aids | Nose & Sinus Clinic, PLLC***", located at:
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2 Sentences were found in a text with the title: „***Understanding Hearing Aid Technologies: Locations in Enid ...***", located at:
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2 Sentences were found in a text with the title: „***Complete Hearing Solutions, VIC | Hearing Loss & Aids ...***", located at:
https://www.completehearingsolutions.com.au/hearing_aid_features.html

2 Sentences were found in a text with the title: „***Clear Choice Hearing Center - Digital vs Analog***", located at:
https://thewoodlandshearingaids.com/digital_vs_analog

2 Sentences were found in a text with the title: „***Listen Clear with iPhone Hearing Aids - Know before you buy...***", located at:
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2 Sentences were found in a text with the title: „***Types of Hearing Aids - Hearing Aids***", located at:
https://www.hearingaid.cc/types-of-hearing-aids/

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http://cs.brown.edu/research/pubs/theses/masters/2009/diamond.pdf

▶ In 118 further documents exactly one sentence was found. (click to toggle view)

Subsequent the examined text extract:

STFT Algorithm for Implementation of Auditory Compensation in Hearing Aids

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Abstract - **The paper presents the work on auditory compensation (also known as audiogram equalizer or loudness compensation) implemented with the help of Short Time Fourier Transform (STFT) algorithm. Auditory compensation is an important feature of the hearing aid. The hearing aid's primary functionality is to compensate the loss in the hearing level of a patient by amplifying the audio signals depending on the frequency band. The amount of amplification required for an individual is determined by the Audiometry test performed by audiologist. The STFT algorithm is employed to determine the frequency range of audio signal which is to be selectively amplified as per the audiogram. The present work uses Verilog language to implement STFT algorithm. The entire system is developed on Zynq Evaluation and Development Board (Zedboard) using Vivado.**

Keywords — **Hearing Aid, Audiogram, Auditory Compensation**

INTRODUCTION

Hearing is one of the important senses which alert us to danger that sometimes may be out of our visual range and speech being the most common form of communication. Sound is produced upon vibration or back and forth movement. Audio or sound is merely quick movement of air molecules which are caused by vibrations caused by such actions. The human ear helps to sense these sounds and transmits the information to the brain.

The human ear is of three parts: inner ear, middle ear and outer ear. The outer ear begins with pinna is structured to gather sounds from different directions and funnel them into the ear canal. The Middle Ear is air filled cavity, it receives sound from external ear auditory canal. The middle ear begins at the inner end of the external auditory canal, specifically at the eardrum. The sound waves sets the eardrum to vibrate, which in turn sets the three tiny bones in the middle ear into motion. The motion of the bones makes the fluid in the inner ear or cochlea to move. The movement of the inner ear fluid causes the hair cells in the cochlea to bend. The hair cells change the movement into electrical pulses. These electrical impulses are transmitted to the hearing nerve and up to the brain, where they are interpreted as sound. So it is important that all the parts of the ear work for us to hear the sound.

Hearing loss or hearing impairment can be partial or total inability to hear, caused by the interruption of audio signals at either or multiple sections of the ear. Certain factors including genetics, aging, infections, exposure to noise, trauma, birth conditions and medications or toxins can invoke hearing loss. Hearing impairment is categorized into: conductive hearing loss, sensorineural hearing loss (SNHL) and mixed hearing loss. Conductive hearing loss is caused by the impediment in conveying the sound in its mechanical form through the middle cavity to the inner ear. Sensorineural hearing loss is due to nerve-related hearing loss. Mixed hearing loss is a combination of the two.

Although there is a wide range of options to treat hearing impairment, hearing aid is one such option in high demand. Hearing aids are sound-amplifying devices designed to increase audibility. Most of the hearing aids are associated with similar components such as microphone, amplifier circuitry, miniature loudspeakers and batteries to power the device. Based on the technology used, hearing aids are classified into: Analog and Digital hearing aid [1]. Analog hearing aids operate by amplifying continuous audio waves. Digital hearing aids convert sound waves into digital signals and produce an exact duplication of sound. The digital hearing aids allow for more complex processing of sound during the amplification process which may improve their performance in certain situations and have greater flexibility in programming so that the sound they transmit can be matched to the needs for a specific pattern of hearing loss.

The Digital Signal Processor block is the heart of digital hearing aid. The main features of the hearing aid depends on the programming of the DSP. The features include auditory compensation, echo cancellation, noise reduction, etc.

Auditory compensation also known as audiogram equalizer is the primary functionality of the hearing aid which is to compensate the loss in the hearing level of a patient by amplifying the audio signals. The work here targets to develop a algorithm for this operation.

Researchers have so far been implementing audiogram equalizer using filter banks for processing audio signal. Multi-channel loudness compensation methods have been achieved in digital hearing aids. For example, a loudness compensation method based on 8-channel filter bank with equal bandwidth was proposed by Thomas [2]. A wavelet transform was introduced by Li M [3]. Audiogram compensation method based on 16 channels non uniform bandwidth filter banks was proposed by Chong K.S [4]. In all these methods, formants of speech are distorted at the junction of filters. So speech intelligibility was reduced. To solve this problem, a multi-channel loudness compensation method based on formant[5] was introduced by Zhao Yi. However, in order to detect formant and redesign filter banks[6], the method has high computational complexity, so that it is rarely used to digital hearing aids.

In the filter bank approach, an array of filters known as analysis filter bank are used to divide the audio signal to different channels based on frequencies. Then an array of amplifier with specific gain is used to amplify signals in each channel. FIR/IIR filters have been used for implementing filter banks. Filter number and the implementation architectures have a significant impact on system performances, such as computation complexity, area, throughput and power consumption. Certain signals are confined by the filters based on their frequency values, therefore the filter component values must be selected effectively else required frequencies may be accidently filtered out.

To overcome the drawbacks of the existing filter bank approach the work proposes a algorithm to use Short-term Fourier transform (STFT) to implement the auditory compensation block.

The algorithm is implemented using Verilog coding and Xilinx Vivado tool is used for writing the code and to run the simulations. The hardware implementation is performed on Zynq-7000 processor. Zedboard has two 12-bit (each) analog to digital converters (ADC) and an in-built Codec – ADAU1761 for analog to digital conversion and audio interfacing with the processor respectively.

auditory compensation

Auditory compensation is the hearing aid's primary functionality to compensate the loss in the hearing level of a patient by amplifying the audio signals depending on the frequency band. The amount of amplification required for an individual is determined by the Audiometry test performed by audiologist.

Each user may have different requirements as per their audiogram results.

Short-term Fourier transform is employed to determine the frequency and phase content of a signal over a period of time. The incoming long term audio signal is divided to short terms of equal length and Fourier transform is computed on each of these shorter segments[9].

The discrete STFT is a time-localized spectral transformation based on the discrete Fourier transform (DFT). The DFT coefficients X(k) of a discrete time signal x(t) composed of T samples are calculated according to

Xk=t=0T-1x(t)e-j2πTkt , k=0,..,T-1

where k is frequency. The DFT is a frequency localized transformation, the analog frequencies equivalent to normalized frequency are fixed and given by

fk =k fsT

where k = 0,1,2,3.....T-1 and f_s is sampling frequency.

The samples of the speech signal are real numbers which makes the DFT to be symmetric.

The STFT can be viewed as a two-dimensional transformation (i.e. frequency and time) which is calculated by splitting the input signal into segments using a sliding time-limited window and then calculating the DFT of each of the segments.

Considering a discrete time input signal, it is segmented into frames according to

xir= wr xr+iD, r=0,….,R-1

where xir is the windowed i-th frame, r is a local time index, R is the window length, and D is the hop size which represents the number of samples that the sliding window moves between two consecutive frames.

The window lengths of the signal may vary from 8, 16, 32 and so on, and the audio frequency is equally divided among these channels. The frequency range to which the audio signal belongs is determined by the position of maximum magnitude of the complex Fourier sequence.

Audio Amplification

Audio amplification is the process of making a small signal bigger by a particular factor without affecting other features of the same. An amplifier which amplifies audio of all frequencies by a same factor is called linear amplifier. In this paper a non-linear amplification method is employed, as the user may require different audio intensity at different frequencies. The user has to undergo an audiogram test to determine his hearing attenuation at various frequencies at both his left and right ear. As per the audiogram result the hearing aid can be configured to amplify the signal to a gain preferred by the user.

III METHODOLOGY

The figure 1 presents the block diagram of the algorithm for auditory compensation algorithm using STFT. The audio signal picked up from the microphone are then converted to digital signals using ADC. The continuous audio signal is divided into smaller segments and STFT is performed on each of these segments. The STFT block finds the FFT of the signal. The FFT implementation is carried out using butterfly method. After FFT calculations, the Max-Bin block calculates the magnitude of each bin of FFT and results with the index of the bin which has got the maximum magnitude. The index represents the frequency of the audio signal. The amplifier block is used to amplify the real time audio signal based on the frequency determined and the amplification requirement data taken from audiogram to compensate the hearing loss of the patient.

ADC

STFT

Max-Bin

Amplifier

DAC

Gain

Figure 1: Block diagram of algorithm for Auditory Compensation

iv. implementation

All the blocks of figure 1 have been implemented using the Xilinx Vivado tool and the hardware platform being Zynq evaluation and development board. Figure 2 represents the schematic of the algorithm implemented.

Figure 2 : Schematic of the Algorithm Implementation.

ADAU1761 codec of Zedboard was utilized to accept audio signal through line in channel. GPIO -1 is the audio input and GPIO-0 is the processed output. Figure 3 is the complete schematic including codec and algorithm. Figure 3: Complete Schematic including Codec and Algorithm

V. RESULTS

The proposed algorithm in this paper has been successfully implemented and derived of satisfactory results.

Implementation involved utilization of certain device resources such as LUT (look up table), FF(Flip-flops), IO (I/O pads), BRAM (Block RAM), MMCM (Mixed-mode Clock manager) etc. Utilization of the resources post synthesis and implementation can be seen in the figure 4.

Figure 4 : Post implementation resource utilization report

Implementation setup includes a Zynq evaluation and development board, Digital storage oscilloscope, microphone and speakers as shown in figure 5. The audio signal of different frequencies were fed as input and the amplified output were observed with the set gain factor.

Figure 5 : Implementation Setup

Vi. conclusion

The advantages of the approach implemented in the work over the filter bank approach are :

In this approach the problem of reconstructing the signal does not exist. Whereas perfect reconstruction of the speech signal is a challenging task in the filter bank approach as the signal is divided into different channels. The resolution in frequency depends on the length N of the STFT whereas in filter bank approach we need to have an additional filter for every channel. The approach solves the problem of Signal Distortion due to filters in filter bank approach. The proposed algorithm in this paper has been successfully implemented and derived of satisfactory results.

VII references

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