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IMPLEMENTATION OF NOVEL ALGORITHM FOR AUDITORY COMPENSATION IN HEARING AIDS USING STFT

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Abstract: Audiogram is the graphic record drawn from the results of hearing tests with an audiometer, which charts the threshold of hearing at various frequencies against sound intensity in decibels. This paper presents the work on auditory compensation (also known as audiogram equalizer) implemented with the help of Short Time Fourier Transform (STFT) algorithm. For humans normal hearing ranges from -10dB to 15dB, although 0dB from 250Hz to 8 kHz is reckoned as the average normal hearing. An audiogram is obtained to determine the frequency range the listener is audibly challenged to. 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: Audiogram, STFT, Zedboard

INTRODUCTION

Hearing is one of the important senses which alert us to danger that sometimes may be out of our visual range. 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 is of three parts: inner ear, middle ear and outer ear. The pinnae are structured to gather sounds from different directions and funnel them into the ear canal. The sound waves are then conveyed onto the middle ear and then the inner ear. 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 cognate 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 (DSP) hearing aid. Analog hearing aids operate by amplifying continuous audio waves. Digital hearing aids come with the similar features of programmable analog hearing aid, but convert the audio signal to digital signal to yield an exact replica of each signal rather than entirely amplifying the signal.

Digital hearing aid alike analog hearing aid, has a microphone to convert the analog audio signals to digital form, a microprocessor to amplify and process the digital signal and a miniature loudspeaker to convey audio directly to the ear canal. The signal processing is executed by the microprocessor in real time taking into account of individual user preferences.

Researchers have so far been implementing audiogram equalizer using filter banks for processing audio signal. 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. Digital sequences have smaller signal bandwidth when compared to analog sequences, which puts signal processing time in a significant position among the factors effecting device performance. FFT algorithm operates at a faster pace and since all signals are considered for evaluation no frequency is filtered out, sanctity of the signal is preserved.

This paper shows the implementation of auditory compensation block by selective amplification of audio signal on Zedboard. Zynq-7000 processor is used to perform the processing and amplification of the signal. 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. The audiogram equalizer is one such algorithm which can be used to customize the hearing aid operation. An individual may have attenuated audibility at a particular frequency range, the STFT algorithm is implemented in this paper to identify the frequency range to be processed and then amplified.

— STFT algorithm

STFT also referred to as 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 [1].

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

$$X(k) = \sum_{t=0}^{T-1} \left(x(t) e^{-j\frac{2\pi}{T}kt} \right), k = 0, .., T-1$$

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

$$f_k = \frac{k\, f_s}{T}$$

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

$$x_i(r) = w(r)\, x(r + iD), \quad r = 0, \dots, R-1$$

where $x_i(r)$ 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. Each channel represents a particular frequency range and as per the user audiogram result.

METHODOLOGY

The figure 1 presents the novel auditory compensation algorithm using STFT algorithm implemented. 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 [2].

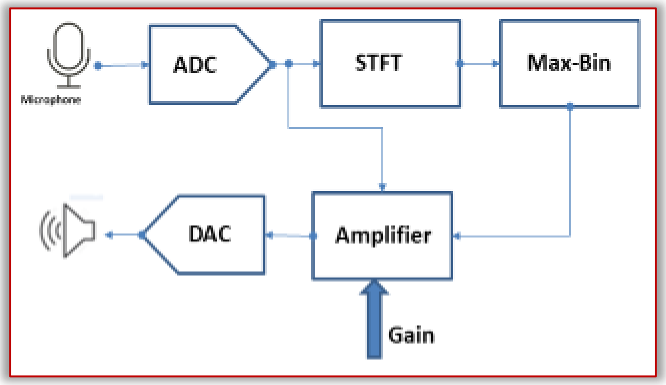


Figure 1: Block diagram of STFT algorithm for Auditory Compensation

IMPLEMENTATION

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

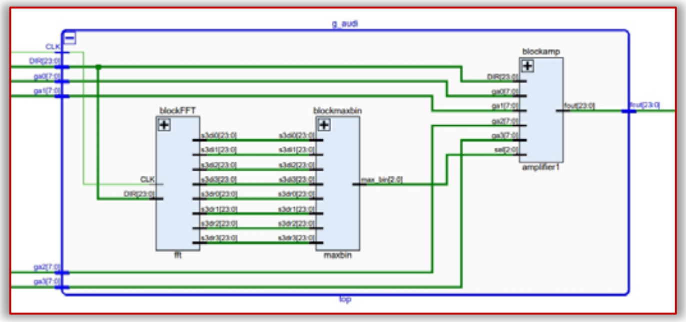


Figure 2: Schematic of the Algorithm Implementation.

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.

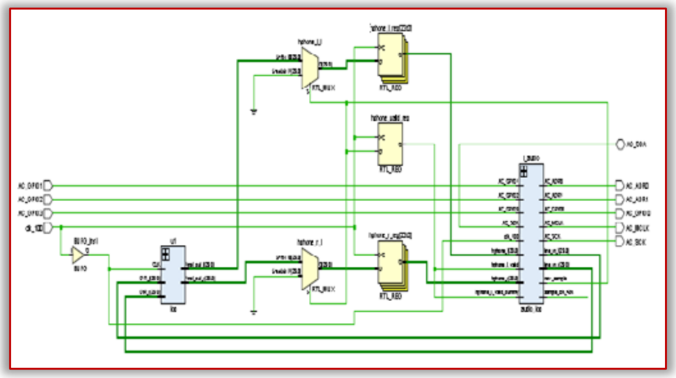
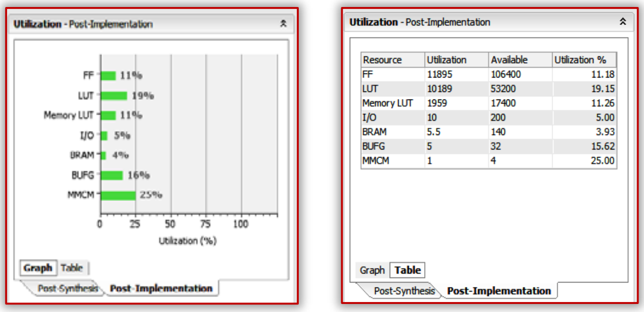


Figure 3: Complete Schematic including Codec and Algorithm

RESULTS

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

Implementation of this project 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.



Resource	Utilization	Available	Utilization %
FF	11895	106400	11.18
LUT	10189	53200	19.15
Memory LUT	1959	17400	11.26
I/O	10	200	5.00
BRAM	5.5	140	3.93
BUFG	5	32	15.62
MMCM	1	4	25.00

Figure 4: Post implementation resource utilization report

Implementation setup includes a Zed 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

CONCLUSIONS

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

- In the novel 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 novel 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.

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