International Journal of Engineering & Technology, 7 (2.33) (2018) 839-842

International Journal of Engineering & Technology

Website: www.sciencepubco.com/index.php/IJET

Research paper

Detection and analysis of symptomatic patterns in audio bio-logical signals at low power consumption and optimized area

Syed Thouheed Ahmed [1] *, Mariyam Fathima [2], Syeda Tasmiya Tarannum [1], Asha Kiran M. U [2]

[1] *Assistant Professor, Department of Computer Science & Engg. Dr. T Thimmaiah Institute of Technology, K.G.F*
[2] *Assistant Professor, Department of Electronics & Communication Engg. Dr. T Thimmaiah Institute of Technology, K.G.F*
**Corresponding author E-mail: syed@drttit.edu.in*

Abstract

Audio signal processing for a biometric analysis of a signal is most challenging field in current era. In this paper, a design is proposed for the acquisition of audio signals and then process under a standard pattern values with the consideration of lower attributes such as power and system scalability for detection of symptomatic pattern. In this paper, a subsequent low power design is used with a mathematical matrix with respect to energy, Quasi-Average (QA), and other various coefficients of wavelets under threshold value analysis. In later process of Mel-frequent analysis is processed and distinguished among signals such as cough and sneeze, with most alike patterns for thresholding value of attribute extracted under matching. The proposed system is more feasible with the user inputs and the Indian environment. Typically, this system is designed and dedicated for Indian digital development

Keywords: *Audio Signal Processing; Biometric; Thresholding; Classification*

1. Introduction

Audio signal processing for a biometric analysis of a signal is most challenging field in current era. In this paper, a design is proposed for the acquisition of audio signals and then process under a standard pattern values with the consideration of lower attributes such as power and system scalability for detection of symptomatic pattern. In this paper a subsequent low power design is used with a mathematical matrix with respect to energy, Quasi-Average(QA), and other various coefficients of wavelets under threshold value analysis. In later process of Mel-frequent analysis is processed and distinguished among signals such as cough and sneeze, with most alike patterns for thresholding, also value of attribute extracted under matching. The proposed system is more feasible with the user inputs and the Indian environment. Typically, this system is designed and dedicated for Indian digital development.

2. Background

"Identification of cough in consistent sound recordings utilizing shrouded Markov models [2]," Cough is a typical side effect of numerous respiratory ailments. The assessment of its power and recurrence of event could give important clinical data in the appraisal of patients with ceaseless hack. In this work of paper [2], the utilization of shrouded Markov models (HMMs) to naturally recognize hack sounds from constant walking soundtracks. The recording framework comprises of an advanced sound recorder and a mouthpiece appended to the patient's trunk. The acknowledgment calculation takes after a catch phrase spotting approach, with hack sounds speaking to the watchwords. This one was prepared on 821 min chosen from 10 walking recordings, including 2473 physically named hack occasions, and tried on a database of nine recordings from particular patients with an aggregate recording time of 3060 min and involving 2155 hack occasions.

The normal identification rate was 82% at a false caution rate of seven occasions/h, while considering just occasions over a vitality edge with respect to each recording's normal vitality. These outcomes recommend that HMMs can be connected to the identification of hack sounds from walking patients. A post processing stage to play out a more point-by-point examination on the identified occasions is being worked on, and could permit the dismissal of a portion of the erroneously recognized occasions. In this broadside [5], the work accomplished is the segments of snore sounds are extracted from audio recorded and further the investigation is carried out and then have concluded whether the subject has suffered from sleep apnea syndrome. However, the system has shown the downside of having less accuracy.

In this broadside [6], a deep survey has been done in order not to criticize, but to serve as allusion intended for the researchers and also to the designers in the arena of scientific research. Overall, it has focused completely on the wearable biosensor systems for the health monitoring systems. The survey says that the power consumption was comparably more and thus concluding it to be unfriendliness, less secure and expensive one.

In the broadside [7] the related work done is with respect to the kids. The proposed system herein has used the wearable sensors and acoustic signal processing in order to provide the monitoring among the children. The disadvantages of the system are only single acoustic signal is processed and consumes more power consumption and is thus less robustness. In the paper [9], speedy progresses were developed to lower the power and to make the system feasible with the numerous wearable and the implementable systems. The system concentrated on symptoms of external

parameters which are naturally produced by humans called acoustic signals. The said designed monitoring system showed the disadvantage of consuming high power.
By considering above all the drawbacks, a system is proposed where three acoustic symptoms belch, burp and cough are considered for detection and analysis. The system is designed in Xilinx 14.7 and thus simulation is done in ModelSim 10.5b. The main focus of the system is to achieve less power in VLSI design

3. System methodology

The primary contribution of the proposed system under design includes single input from the human audio recording with respect to the patient disease pattern and further is to have internal comparison of threshold value, using threshold range table as shown in Table 1: And A Detailed Flow Design is Shown in Fig 1.

Table 1: Mapping Frequency for Threshold

Dwt Coefficient	Frequency Range
D1	2.5- 5.5 Khz
D2	1375 Hz-2.75khz
D3	687 Hz-1375 Hz
D4	343-687 Hz
D5	172.5 Hz-343 Hz
D6	86.25 Hz-172.5 Hz

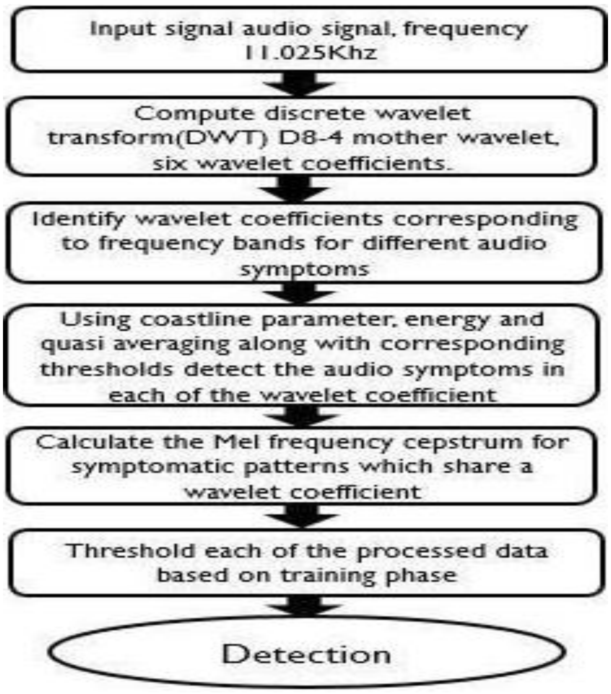

Fig. 1: Proposed Methodology.

The paper proposes a systematic mathematical system for designing and development of biomedical audio signal to detect symptomatic patterns in non-speech processing. These include audio recording such as cough, burp and belch. These signals are considered to be more similar and have commonly observed signal patterns. In Kids these ranges of threshold pattern internally match the system and create a ranging anemology for detection and decision making. In order to create a reliable and wearable sensor design, a most suitable pattern of power consumption is supposed to be proposed. Thus a successful design is possible by optimizing the algorithmic efficiency and power efficiency under the design process. Each component in the system is designed and studied under a specific pattern. In the design process, the system utilizes mathematical matric such as average and coastline parameters. Using such patterns, the classification is made simpler and more dominating to that of the previous approach.

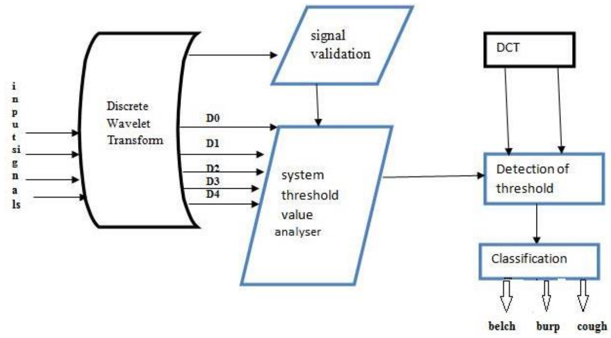

Fig. 2: Block Diagram for Proposed System.

The proposed system in Fig(a) demonstrates the study alignment of the components in a mathematical approach cum optimized manner for the retrieval of human recorded signals, the input is primarily feed to the DWT (Discrete Wavelet Transformation) Unit for internal conversion of HEX signals feed from as input signal cum audio. The biomedical signals are then retrieved from the DWT blocks and compared with the internal thresholding ratios from D1 to D6 in a subsequent manner of ranges for a particular audio biomedical signal.
In the regards, the system is collaborated with internal clock for stabilizing the functionalities of the system internal operations and to maintain the signal originality at thresholding ratio. The system is aided with Mel-Filters for de-noising and signal strengthen for performing higher ratio of optimization. The outcomes of Mel-Filters are layered according to energy spectrum for avoiding internal anomalies of signal. Discrete Cosine Transformation is performed for energy spectrum for internal segregation of the signal strength and in general, the signals are appending with decision making approaches.

3.1. Discrete wavelet transformation (DWT)

For resolving signals into multiple operational signals and then restoring it's temporal and spectral information with internal justification to be for an ideal choice for spectral resolution as compared to the Fast Fourier Transform (FFT) and STFT (Short Fourier Transform) and others. The major goal of DWT is to resolve major patterns into narrow frequency ranges and wavelet coefficient. The various values of D_i are repowered with splitting into ranging values from D_1 to D_6 where each coefficient resembles the attribute of biomedical symptom such as vomit, wheezing, burp, belch, cough and sneezing etc. the same is discussed in Table 1.
In reference to DWT, Mallat's algorithm is utilized in this proposed system for development of optimized model. It typically consists of a low pass filter (H_) and High pass filter (G_) with a cascading sampling as shown in Fig 4.2 which is accomplished by proper checking of the channels in progressive stages. The quantity of channel phases in the DWT piece relies on upon the amount of coefficients of enthusiasm for the framework. Thus, six falling phases of H and G are required. In the meantime, the five acoustic examples are recognized utilizing the wavelet coefficients D3 through D6, so we need five H channels and four G channels. Every one of these channels are of the [8] request because of the utilization of the Daubechies fourth-arrange mother wavelet. A standard execution of [9] channels of the eighth request would be computationally escalated as far as number of augmentation. We use multiplier-less system of CSHM (calculation sharing multiplier) and normal sub-expression elimination (CSE) to diminish control utilization.

3.2. Mathematical metric block

The mathematical metric is as shown in fig 4.3 below with internal operation of Di values. The block computes the energy parameters since it consists of multiplier and accumulator, which adds square value to the D6 wavelet coefficient. The average energy value is then compared to the threshold to sense the concerning data patterns. The CL parameters are calculated based on the D5 wavelet coefficient. If the two input samples are in similar ratio, the clock is delayed with a cycle for analyzing the internal thresholding. The magnitude of the differences is acuminated over a range of pixel windows recomputed the comparison ratio with skipped clocks. Hence on comparison with threshold value, this is considered as D5 signal ratio rather than wheezing.

3.3. Quasi average for burp and belch

The decomposition ratio of burp and belch is considerably similar to the patterns considered in the overall system. The D4 and D5 provide a justification of these two audio input signals with respect to the thresholding values range. The signal detection at this stage is challenge and thus to provide a reliable format of decision, wavelet coefficient with appropriate weights is considered and computed.

$$w_{K+1} = \frac{1}{W} s_{i:i+w} - < W_k > + x_{i+w+} \quad (1)$$

Subsequently, a calculation level alteration was utilized called Quasi Average (QA). QA is registered via altering the customary meaning of normal thru a supposition that every component of a consistently moving window is genuinely spoken to by the normal estimation of its window. Such a guess brings about a satisfactory blunder of the request of 10 −6. (1) is the numerical portrayal of QA.

3.4. Discrete cosine transformation (DCT)

A key segment of the JPEG Image Compression Standard is the change step. The objective of this progression is to move (change) the preprocessed picture to a setting where the coding segment of the pressure calculation can be more viable. The coding strategy works best if there are generally couple of unmistakable qualities. We require a change that endeavors this perception we look for a change that takes a N×M square contained comparable values and maps the piece to a network of a similar size where a large portion of the data about the first piece is put away in generally couple of components and the rest of the components are either zero or near zero. On the off chance that each district of like qualities is mapped to values almost zero, then the yield of the change will comprise of a generally extensive measure of close to zero qualities.

The most recent decade, Discrete Cosine Transform (DCT) has developed as the accepted picture change in most visual frameworks. DCT has been generally sent by current video coding models, for instance, MPEG, JVT and so forth. This report presents the DCT, expounds its imperative qualities and breaks down its execution utilizing data theoretic measures.

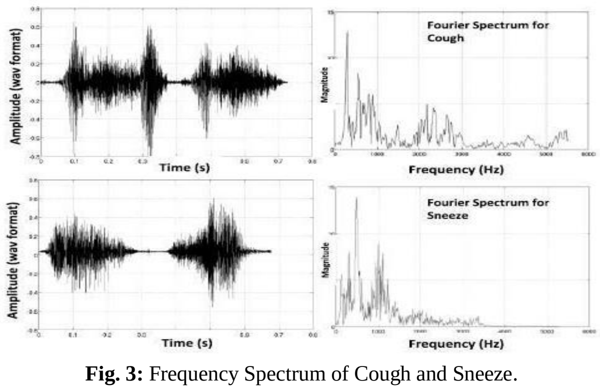

Fig. 3: Frequency Spectrum of Cough and Sneeze.

3.5. MEL-filters

Psychophysical considers have exhibited that human impression of the repeat substance of sounds, either for faultless tones or for talk signals, does not take after a straight scale. Characterizing subjective pitch of faultless tones has been gotten from this examination. How high the voice is i.e., its pitch, is a mental estimation that addresses how high and low sounds are. Pitch is hugely, yet not just, dependent on repeat; the energy of sounds similarly impacts the impression of pitch. The melodic size of pitch is one measure that addresses the mental measure of pitch; however octaves and increasing the pitch are not a comparative sensation. With a particular ultimate objective to build up a mental measure for different assessment of the pitch sensation we need to do contribute estimation tests light of the size estimation procedure like those of a couple of estimations for mayhem for making sense of what is heard is twofold the pitch and what is heard as a vast bit of the pitch of a standard sound. One mental measure of contribute made along these lines is the Mel scale. The Fig 4.5 shows the frequency spectrum of the cough and sneeze signals which will be falling in the same wavelet coefficients, thus this Mel frequency Cepstrum coefficient based analysis will be further applied to the signal after DWT operation to distinguish between these two signals. The Mel scale (in KHz) can be approximated by taking after condition as shown in Equation

$$\text{Mel}(f) = 2595 \log_{10}(1+f/700) \quad (2)$$

The figure 4.5 shows the graph of Mel scale versus fr quency scale and the arrangement stream of Mel filter bank is showed up in figure 4.6. Thirteen Fir filters, illustrated with Bartlett window, are used, having assorted low pass and high pass frequencies. The arrangement as showed up in figure 8 is checked using three differing course of action of repeat filter bank to make a comparable audit among all. Starting now and into the foreseeable future, the yield of each filter is experienced full snake to get the yield coefficients of MEL filter bank.

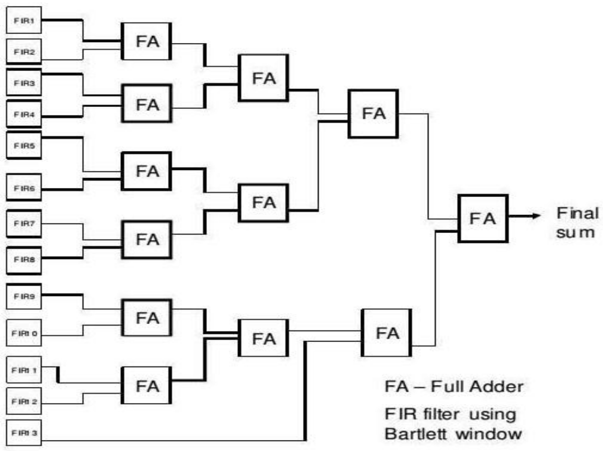

Fig. 4: Architecture of Mel Filter Bank.

4. Observative analysis

According the design of audio signal processing of the given biomedical human recorded signal, the methodology is designed as follows, in the primary step input is collected under digital form with an incoming frequency of 11KHz and 10-bit resolution. On successfully retrieving the signal, the DWT is performed for wavelet coefficient extraction from D1 to D6 under a threshold value table range. On extraction, the identification of the patterns is considered and provided with corresponding frequency band. On similarity indexing of two thresholding value signal is also segregated. The table 1 below displays the classification of threshold value on input signals. Using the Mel Filters, the relationship coefficient is computed and fetched for detailed segregation of the matched signals and the decision making is provided under a DCT confirmation with energy spectrum. The Table 2 shows the classification of threshold value on the input signals.

The designed proposed system is developed in Xilinx package for internal feasibility and fine-grained in ModelSim, an Intel package for signal pattern segmentation and wavelet analysis.

Table 2: Classification Table for Signal in Proposed System

Input Pattern	Wavelet Coefficient of Interest	Processing Method	Percentage Classified		
			Cough	Sneeze	Burp/Belch
Cough	D_3	MFCC based	**90.3 %**	3.22%	6.45%
Sneeze	D_3	MFCC based	16.7%	**84.3%**	0%
Burp/Belch	D_4D_5	Quasi-average	0%	0%	**100%**

4.1. Register transfer level

Fig 5 describes RTL view of the designed system with respect to the incoming signals and the ratio of pins and design. The shown figure is a detailed representation of audio signal block system. The logic synthesis tool will produce the gate level netlist commencing the RTL level design. The Top Module of the designed system consists of four inputs namely clk, reset, start, daudio_input (7:0) and three outputs which are belch, burp and Cough.

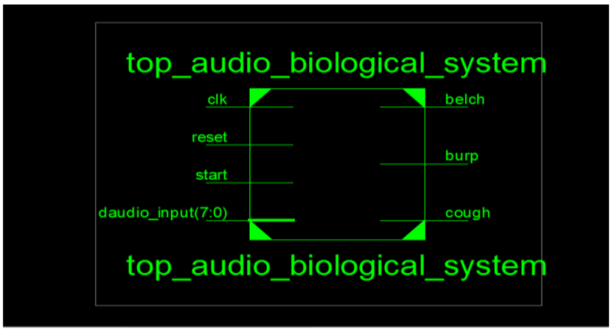

Fig. 5: RTL View of Proposed Block.

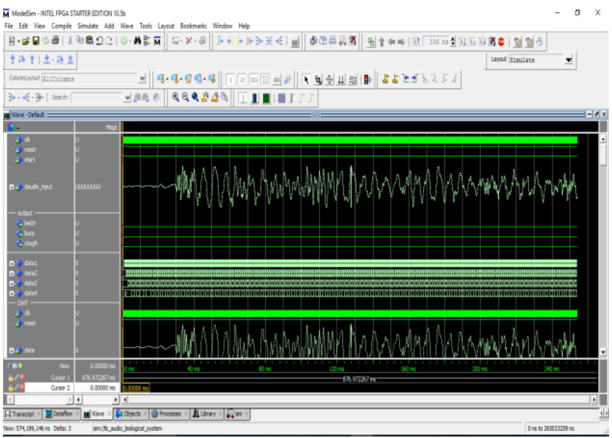

Fig. 6: Signal Processing for Data Sample (Cough).

The proposed system is designed based on the threshold value range analysis using Mel Fitters, DCT and DWT for the decision making. The design of proposed system is in Xilinx tool with ModelSim for fine graining overall wavelet coefficients. In the proposed system, we have analyzed three input samples and its classification based on internal thresholding as burp belch and cough under Indian ecosystem using minimal filtering ratios of Mel Filters. The outcomes are accurate and the coefficient comparison is successfully provided for matching. The system has now classified signal ratios and provided minimal or optimized range of computation and processing.

As discussed, the system is solely designed for the collection of recorded signals it can be improvised to retrieve a dynamic signal collection and pattern generation for disease analysis. These signals are threshold and hence in near future, machine learning approaches can be appended for self-learning and knowledge mining of incoming data patterns.

5. Conclusion

The proposed system is designed based on the threshold value range analysis using Mel Fitters, DCT and DWT for the decision making. The design of proposed system is in Xilinx tool with ModelSim for fine graining overall wavelet coefficients. In the proposed system, we have analyzed three input samples and its classification based on internal thresholding as burp belch and cough under Indian ecosystem using minimal filtering ratios of Mel Filters. The outcomes are accurate and the coefficient comparison is successfully provided for matching. The system has now classified signal ratios and provided minimal or optimized range of computation and processing.

As discussed, the system is solely designed for the collection of recorded signals it can be improvised to retrieve a dynamic signal collection and pattern generation for disease analysis. These signals are threshold and hence in near future, machine learning approaches can be appended for self-learning and knowledge mining of incoming data patterns.

References

[1] S. Matos, S. S. Birring, I. D. Pavord, and D. H. Evans, "Detection of cough signals in continuous audio recordings using hidden Markov models," IEEE Trans. Biomed. Eng., vol. 53, no. 6, pp. 1078–1083, Jun. 2006.

[2] A. Pantelopoulos and N. G. Bourbakis, "A survey on wearable sensorbased systems for health monitoring and prognosis," *IEEE Trans. Syst., Man, Cybern. C, Appl. Rev.*, vol. 40, no. 1, pp. 1–12, Jan. 2010.

[3] A. Basak, S. Narasimhan, and S. Bhunia, "KiMS: Kids' health monitoring system at day-care centers using wearable sensors and vocabularybased acoustic signal processing," in *Proc. 13th IEEE Int. Conf. e-Health Netw. Appl. Services (Healthcom)*, Jun. 2011, pp. 1–8.

[4] W.-H. Liao and Y.-K. Lin, "Classification of non-speech human sounds: Feature selection and snoring sound analysis," in *Proc. IEEE Int. Conf. Syst., Man Cybern.*, Oct. 2009, pp. 2695–2700.

[5] Low-Power System for Detection of Symptomatic Patterns in Audio Biological Signals,Himanshu S, Markandeya and Kaushik Roy, Fellow, IEEE transaction 2016.

[6] V.Swarnkar, U R Aberatne,YusufAmrulloh, Craig Hukins, RinaTriasih, AmaliaSetyati, IEEE 2013"Neural Network based algorithm for automatic identification of cough sounds".

[7] S. B. Davis and P. Mermelstein, "Comparison of parametric representations for monosyllabic word recognition in continuously spoken sentences," IEEE Trans. Acoust., Speech, Signal Process., vol. 28, no. 4, pp. 357–366, Aug. 1980.

[8] F. Zheng, G. Zhang, and Z. Song, "Comparison of different implementations of MFCC," J. Comput. Sci. Technol., vol. 16, no. 6, pp. 582–589, Nov. 2001.

[9] SSouani, M. Abid, K. Torki, and R. Tourki, "VLSI design of 1-D DWT architecture with parallel filters," Integr. VLSI J., vol. 29, no. 2, pp. 181–207, 2000.

[10] Syed Thouheed Ahmed S and Kiran Kumari Patil "Digital Telemammography Services for rural India, software components and design protocol" Proc. IEEE International Conference on Advances in Electronics, computers and Communications, 2014, pp 1-5

[11] Syed Thouheed Ahmed "A Study on Multi Objective optimal technique for medical datasets" Proc. IEEE International Conference of Intelligent Computing and Control Systems, 2017, pp 174-177.