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Design of a Low Power and Area Efficient Architecture for the Detection of Audio Biological Signals

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Abstract: In digital audio recording the audio signals are picked by a microphone or other transducer and converted into a stream of discrete numbers, representing the changes over time in air pressure for audio, then recorded to a storage device. Hence these audio recordings are analyzed to detect the audio biological symptoms, such as cough, sneeze, vomiting, wheezing, belching and so on, which are spectrally analyzed using a discrete wavelet transform (DWT). The DWT will help to find out the signal level of variation and also use simple mathematical metrics, such as energy, quasi-average, and coastline parameters. These parameters are used to find out type of symptomatic patterns to be detected. Furthermore a Mel-frequency cepstrum-based analysis is applied to distinguish between signals, such as cough and sneeze, which have similar frequency response and hence occur in common wavelet coefficients. The proposed approach is to detect the symptomatic patterns using acoustic non speech human signals which increases the efficiency of mathematical metrics and in particular reduces the area occupied by the architecture. Thus the aim of the proposed work is to design a low power and area efficient mathematical architecture for the calculation of energy parameter, coastline parameter, quasi-average and Mel Cepstrum based analysis for the detection of different symptomatic patterns in audio biological signals. Existing method uses Binary Common Sub-Expression Elimination (BCSE) technique in the design of multiplier which is used in the architecture of DWT and energy parameter calculation. The proposed work employs Multiple Constant Multiplier (MCM) technique in the design of DWT and energy parameter calculation. The design based on MCM outperforms the design using common sub-expression elimination technique with respect to hardware resources and power consumption thereby making the design a low power and area efficient.
Keywords: Audio Biological Signals, Mathematical Metrics, Symptomatic Patterns, Mel Cepstrum,Low Power.

I. INTRODUCTION

Digital recording is the audio signals picked up by a microphone or other transducer or video signals picked up by a camera or similar device which are converted into a stream of discrete numbers, representing the changes over time in air pressure for audio, chroma and luminance values for video, then recorded to a storage device. Audio signal processing or audio processing is the intentional alteration of audio signals often through an audio effect or effects unit. As audio signals may be electronically represented in either digital or analog format, signal processing may occur in either domain. Analog processors operate directly on the electrical signal, while digital processors operate mathematically on the digital representation of that signal. Analog Signal Processing (ASP) then involves physically altering the continuous signal by changing the voltage or current or charge via various electrical means. A digital representation expresses the pressure wave-form as a sequence of symbols, usually binary numbers. Processing methods and application areas include storage, level compression, data compression, transmission, enhancement (e.g., equalization, filtering, noise cancellation, echo or reverb removal or addition, etc.) Health monitoring is the continued oversight of the progression of a clinical trial. This is to ensure that it is conducted according to protocol as well as good clinical practice, regulatory requirements and standard operating procedures. The purpose of monitoring is to see whether a particular intended result or set of results has actually happened after a clinical process or substance has been applied and to provide ongoing oversight to the quality of care given to meet a person's need. In Literature [1] proposed a generic system based on wavelet transform, mathematical metrics, and mel cepstrum based analysis, which can be used to detect symptomatic patterns in audio biological signals. Modifications in the algorithm and the use of low-power methodologies to implement the algorithm into circuit enable the design of a low-power system. The system can be scaled to include other health markers and can also be made user-specific.but the drawback was that it Consumes more number of hardware resources.[2] proposed architecture which employs floating-point arithmetic operations to minimize the operation bit-width and the total size of LUTs. Furthermore, a floating-point MAC unit and memories are shared with many processes to reduce hardware complexity and energy consumption remarkably but at the cost of operating Speed.[3] and [4] have presented a new method for non-intrusive quality assessment of noise-suppressed speech, by using mel-filter bank energies as features to capture signal variations, and SVR for feature mapping. We showed that noise injection and suppression affects the FBEs and such changes (represented by the mean and variances) are also effective and parameterizable to assess quality. But this leads to a complex and time consuming task. [5] proposed a work which attempts to comprehensively review the current research and development on wearable biosensor systems for health monitoring to evaluate the maturity level of the top current achievements in wearable health monitoring systems. A set of significant features, that best describe the functionality and the characteristics of the system has been selected to derive a thorough study. This system can detect only a single acoustic symptom Session II gives a detailed description about each block available in the audio biological system. Session III elaborates on the existing work and the proposed work of the thesis.session IV describes the results and explanation of the obtained results. And also the simulation tools used to simulate the designed audio biological system. Session V concludes the project and also highlights the scope for the future work.

II. AUDIO BIOLOGICAL SYSTEM

Technology scaling has resulted in the development of novel applications in a wide array of fields. The field of medical systems is no exception to this and has benefitted immensely. In the past decade, rapid advancements in the development of low-power design methodologies have resulted in feasible designs for various wearable and implantable medical systems. Numerous wearable health monitoring systems have been proposed in order to deliver early warning of an impending health condition. These systems monitor various internal as well as external parameters related to the human health, such as temperature, heart rate, and so on. Apart from these parameters, it is well known that acoustic symptoms, such as cough, sneeze, belching, and so on, are early markers of serious health issues, such as influenza, diarrhea, and whooping cough, especially among children. If repetitive occurrence of these symptoms is detected in advance, it is possible for the patient or the healthcare personnel to commence remedial action prior to aggravation of the problem.

An algorithm has to be derived and its corresponding circuit to detect symptomatic patterns in human acoustic non-speech signals. These include audio recordings of cough, sneeze, belch, wheeze, and vomit patterns. These five human non-speech audio tracks are selected, because they are the most commonly observed signals. They are also known to be symptoms for diseases ranging from influenza, ear infection to serious conditions, such as asthma, bronchitis, stomach flu, and so on. It should be noted that apart from the identified five acoustic symptoms, the proposed system is scalable to other human non-speech audio as well. In order to correctly classify the type of symptom, the acoustic signal needs to be processed efficiently to cause detection. Complexity of this processing is directly translated into equivalent power consumption of corresponding hardware implemented. In order to design an effective and long lasting wearable system for symptomatic pattern detection, it is necessary to reduce its power consumption without degrading the efficacy of detection.

A. Architecture Of Audio Biological System

A successful design can be achieved by optimizing algorithmic efficacy and hardware power efficiency during the design process. Previously, such approach has been used in the development of implantable systems as well. Using intelligent approximations at the algorithm level and low power circuit techniques, it was shown that a high efficacy of pattern detection can be achieved while maintaining power efficiency.

The primary contribution is to address two important issues. First, using a single input (human audio recording), multiple symptomatic patterns have been identified with a high efficacy. Second, the implemented hardware has been made scalable over variety of signals and power efficient.

Methodology can be extended to efficaciously detect other symptomatic patterns using power-efficient circuits. Using the wavelet transform as a mathematical tool to resolve the acoustic signals into their spectral components. Each component can be subsequently identified for specific pattern.

In order to reduce the effect of sporadic spikes and noise in the signal, we have utilized the statistical nature of mathematical metrics, such as average, coastline (CL), and so on. Using such methods, the dominant patterns can be detected and classified efficaciously. Furthermore, we have used processing based on mel cestrum computation to detect signals, which have indistinguishable frequency spectrum.

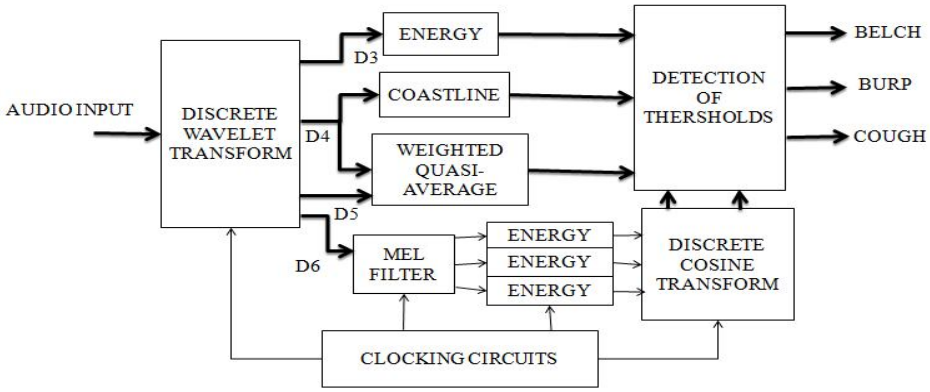

Figure 1. Basic Architecture for the Detection of Audio Biological Signals.

Using low-power design methodologies, such as multiplierless filters, the power constraints on the design are met. The Basic Architecture for the Detection of Audio Biological Signals as shown in figure 1.This enhances the feasibility of integrating the system into a wearable product. Design parameter choices have been made at algorithm and circuit levels of abstraction in order to achieve power efficiency in the implementation. For instance, ideally, a wavelet transform would be sufficient to decompose a signal into its component frequencies. However, to do that at a lower hardware cost, we make modification to filter coefficients (algorithm modification) and filter circuit topology (circuit modification) to achieve similar functionality without any degradation in quality at a much lower hardware cost and power.

B. Blocks of the Audio Biological System

For a wearable product, it is necessary to optimize power consumption with functional efficacy. Hence, optimal signal processing techniques need to be selected depending on signal analyzed, hardware cost and computational efficiency. Several mathematical tools, such as fast Fourier transform (FFT), short-time Fourier transform (STFT), wavelet transform, and so on, can be used to spectrally analyze the acoustic. Another technique that can be used to analyze audio signals is the S-transform. This is an extension of continuous wavelet transform, where the STFT is calculated over a window of varying width. This gives a better resolution of the signal. However a universal system uses the algorithm-circuit co-design approach for detecting multiple symptomatic patterns from a single input acoustic signal. These symptoms have specific frequency composition corresponding to each pattern. The audio signal is streamed, and it is essential to preserve the spectral as well as the temporal information in the signal, which can be achieved by using wavelet transform. The frequency-resolved signal is subsequently processed using mathematical metrics and mel cepstrum-based analysis in order to cause detection.

The main blocks of the audio biological system includes the Discrete Wavelet Transform (DWT), Mel-filters, Discrete Cosine Transform (DCT) and the mathematical metrics such as energy parameter, coastline and quasi-average. This section provides the detailed description about each blocks.

C. Discrete Wavelet Transform

Discrete wavelet transform (DWT) is a common signal processing tool used for multi-resolution analysis of various types of signals. DWT decomposes the input signal into narrow bands of its component frequencies. The wavelet transform block is the most computationally intensive block in the system and consumes a significant amount of power. There are various methods available in the literature to implement the DWT block. The DWT block consists of consecutive stages of low-pass (H) and high-pass (G) filters. These cascading stages are separated by intermediate sub-sampling, which is achieved by appropriate clocking of the filters in successive stages. The number of filter stages in the DWT block depends on the number of coefficients Decomposition is represented in the form of approximate and detail coefficients. While the approximate coefficients correspond to the low-frequency/coarser variations of the signal, the detail coefficients are the high frequency/finer variations. DWT uses various types of wavelet and scaling function as the basis for signal decomposition. Choosing an appropriate wavelet function is essential for an accurate resolution of the signal. Due to multi-resolution property, DWT helps in preserving both spectral and temporal information in the signal unlike FFT. It also has a better resolution as compared with STFT due to dyadic scaling. Traditionally, wavelet transform has been used extensively in image processing, especially for applications requiring data compression. In recent times, it has also been used in analyzing biological signals in field of bioinformatics and neuroscience. Apart from the above-mentioned advantages of using DWT, the hardware implementation of DWT using techniques, such as Mallat's algorithm or lifting facilitates low-power design. The proposed system has to distinguish and segregate the five acoustic signals efficiently. Due to this requirement and the advantages over FFT/STFT, DWT is selected for the spectral resolution of input signals.

The Mallat's algorithm, used to implement the wavelet transform, uses lower order filters in combination with sub-sampling operation to resolve the signal into very narrow frequency bands. This is advantageous in implementing the hardware. However, the wavelet resolved signal needs to be processed further in order to remove the sporadic spikes and noise, which might trigger a false detection.

D. Mel-filters

Mel frequency cepstrum coefficients (MFCCs) are the set of coefficients extracted from an audio signal. It is extensively used in speech or speaker recognition. The principle of MFCC is based on the fact that the sounds generated by human vocal tract are modulated by the shape of the tract, including the tongue and teeth. The shape is manifested in the form of an envelope of power spectrum over short periods of time. MFCC have been shown to accurately represent this envelope. The primary component is the mel filter bank, a set of overlapping bandpass filters which are uniformly spread around the center frequency on the mel scale.

The formula for converting from frequency to Mel scale is:

$$M(f) = 1125 \ln(1 + \frac{f}{700}) \quad (1)$$

To go from Mels back to frequency:

$$M^{-1}(m) = 700(\exp\left(\frac{m}{1125}\right) - 1) \quad (2)$$

The mel scale describes the human auditory system on a linear scale. The conversion between mel scale (m in mels) and frequency scale (f in Hz) is computed using (1) and (2). Based on the uniformly spread center frequencies of the mel filters transform to a logarithmic spacing on the frequency scale. This transformation is coherent with the fact that the human cochlea cannot discern the difference between two closely spaced frequencies, especially at higher frequencies. The mel scale assumes an almost linear transfer of power for frequencies under 1 kHz and a logarithmic dependence for higher frequencies, thereby mimicking the human auditory system. The spectral energy in each filter in the mel filter bank is then given to the logarithm block for a nonlinear normalization. Since the filters are overlapped, there is significant correlation between the spectral filter energies.

Thirteen FIR filters, designed with Bartlett window, are used, having different low pass and high pass frequencies. The MFCC-based analysis uses the DWT as the first stage spectrum instead of the FFT. The number of mel filters in the mel filter bank is reduced due to the resolution of the cough and sneeze signals into a single coefficient of interest. Three overlapping bandpass filters are used in the mel filter bank. It can be observed from the mel filter bank response in Figure 3.

The mel filters are designed for a triangular magnitude response around the center frequency. These filters are of the 16th order, so that the frequency response is closely matching the required triangular response. The coefficients of these filters are adjusted by reducing the number of 1s. This reduces the number of computations without adversely affecting the frequency response of the filter.

E. Mathematical Metric Blocks

The block diagrams for the mathematical metric blocks which induces for the detection of audio biological signals by detecting the threshold values for each biological signals. It consists of a multiply and accumulate operation, which adds the squared value of the input. The input window size is chosen in the training phase and corresponds to 1024 samples of the digitized input data.

$$E_{AVG}[n] = \frac{1}{N}\sum_{i=1}^{N} E\,(i + (n-1) * N) \quad (3)$$

The average energy value is then compared against the threshold to detect acoustics pertaining to belching sound. Energy parameter captures the continuous increase in the amplitude of the low-frequency component in human auditory signal to correctly detect this symptom.

The CL parameter is calculated based on (4).

$$CL(k) = \sum_{i=1}^{N} x[i + (k-1) * N] - x[i - (k-1) * N] \quad (4)$$

where x is the input data and N is the window size for kth window.The input is delayed by a clock cycle in order to calculate the difference between two adjacent samples. The magnitude of the difference is accumulated over a prefixed window in order to calculate the trace length of the signal. This accumulated value is then compared with the threshold for detecting coughing. Since coughing signal is periodic signal for time duration without any significant increase in amplitude, the CL parameter captures this pattern accurately.

In order to enable a memoryless implementation and a continuously moving average, the average calculated in the previous window is subtracted from the sum of the running window instead of the individual data sample. Since the window size is a power of two, the divider is implemented by discarding the appropriate least significant bits. The weights are used to normalize the magnitudes of the two coefficients. The weighted sum is compared with a prefixed threshold to detect occurrence of belching or burping pattern.

$$\langle W_{k+1} \rangle = \frac{1}{w}(S_{i:i+w} - \langle W_k \rangle + x_{i+w+1}) \quad (5)$$

where W is QA of kth window, S is the accumulated sum of the k^{th} window, and w is the window size.

F. Discrete Cosine Transform

Discrete Cosine Transform (DCT) is a mathematical tool that has a lot of electronics applications, from audio filters to video compression hardware. DCT transforms the information from the time or space domains to the frequency domain, such that other tools and transmission media can be run or used more efficiently to reach application goals: compact representation, fast transmission, memory savings, and so on.

DCT approaches the statistically optimal transforms for highly correlated signals, it is widely used in digital signal processing, especially for speech and image data compression. Thus many algorithms and VLSI architectures for the fast computation of DCT have been proposed. For the fast computation of 2-D DCT, the conventional approach is the row-column method. This method requires 2N I-D DCT's for the computation of the N x N DCT. However, for hardware parallel implementation of the conventional approach, a complicated matrix transposition architecture as well as 2N 1-D DCT modules is required. Thus for more efficient computation or parallel implementation of the 2-D DCT, the algorithms that work directly on the 2-D data set has to be introduced.

The DCT block is also designed by modifying the coefficient matrix in order to reduce the number of 1s and facilitate the CSHM-based implementation. The first output coefficient of DCT corresponds to the dc component and can be ignored. The second and third coefficients correspond to the cough and sneeze patterns, respectively. These MFCC-based parameters are then compared with a threshold, predetermined during training to detect the corresponding pattern. Depending on the type of signals being used, it is possible to modify the number of the mel filters in the filter banks and correspondingly modify the hardware implementation to achieve scalability and patient specific programmability.

In this chapter the basic architecture for the detection of audio biological signals is discussed along with the brief functionality of each block of the audio biological system. This helps in carrying the uses of each block throughout the process which proposes for the detection of the biological signal. And also the detailed architecture of every blocks of the audio biological system is studied.

III. MULTIPLIER TECHNIQUE

In the design of systems using digital signal processing and other applications multiplier is an important basic building block. Many researchers are continuously trying to design multiplier with high speed, low power consumption, regular structure, such that it occupies less area for compact VLSI implementation. Many algorithms are proposed in the past to perform multiplication process. Every algorithm offerings its own advantages and having tradeoff between themselves by means of their speed area, power consumption and circuit complexity.

A. Existing System

A standard implementation of nine filters of the eighth order would be computationally intensive in terms of number of multiplication. The existing method utilize multiplier technique of computation sharing multiplier (CSHM) and common sub-expression elimination (CSE) to reduce power consumption. These are well-known low-power methodologies, where the filter coefficients are represented using the minimum number of alphabets and their pre-computed products with the input data. The partial products of the input data with the filter coefficients are subsequently computed by shifting and adding these pre-computed

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products and reusing the intermediate sum. The choice of filter coefficients (algorithm level) and multiplier filter (circuit level) applies the algorithm circuit co-design approach. The wavelet coefficients are normalized before subsequent processing to reduce data path width and maintain the correlation.

B. Common Sub-expression Elimination

FIR filter has wide application as the key component in any digital signal processing, image and video processing, wireless communication, and biomedical signal processing systems. Moreover, systems like Software Defined Radio (SDR) and multi-standard video codec need a reconfigurable FIR filter with dynamically programmable filter coefficients, interpolation factors and lengths which may vary according to the specification of different standards in a portable computing platform. Significant applicability of an efficient reconfigurable fir filter motivates the system designer to develop the chip with low cost, power, and area along with the capability to operate at very high speed. In any fir filter, the multiplier is the major constraint which defines the performance of the desired filter. Therefore, over the Past three decades, design of an efficient hardware architecture For fixed point fir filter has been considered as the major research focus as reported.

The number of adders (subtractors) termed as Logic Operators (LOs), used to implement the coefficient multiplications determines the complexity of FIR filters. Hence, the methods that minimize the complexity of multiplication in FIR filters focus on reducing the number of LOs. Among the approaches for reducing the LOs, the CSE technique produced considerable reduction of LOs. The aim of CSE is to identify multiple occurrences of CSs that are present in the coefficient set and to eliminate the redundant multiplications. As the computation of multiple identical expressions needs to be implemented only once, the amount of hardware used can be reduced. This in turn reduces the area and power of FIR filters. Various methods, which utilize the CSs that occur in the CSD representation of filter coefficients. A CSE method based on Binary representation of coefficients, which produces better reduction of LOs compared to the CSD-based CSE methods. The reduction of LOs was obtained by combining three techniques- the 3 bit binary Horizontal CSE (HCSE), Vertical CSE (VCSE) and hardwiring of final stage adders. BSE (Binary Sub-expression Elimination) reported that 4-bit/higher order CSs may not reduce LOs significantly. A 4-bit BCSE (Binary Common Sub-expression Elimination) that offers a better LO reduction. The 4-bit BCSE comprises of HSCE along with Horizontal Super Sub-expression Elimination (HSSE). HSSE can be formed by exploiting redundant identical shift between Horizontal CSs (HCSs).

Binary common sub-expression elimination (BCSE) algorithm is one of those techniques, which introduces the concept of eliminating the common sub-expression in binary form for designing an efficient constant multiplier, and is thus applicable for reconfigurable FIR filters with low complexity. However, the choice of the length of the binary common sub-expressions (BCSs) makes the design inefficient by increasing the adder step and the hardware cost.

C. Proposed System

To design a power and resource efficient architecture for the detection of audio biological signals, the proposed work aims at modifying the architecture of DWT using multiple constant multiplier. To design a low power and area efficient mathematical architecture for the calculation of energy parameter based analysis for the detection of different symptomatic patterns in audio biological signals.

D. Multiple Constant Multiplier

Multiple Constant Multiplication (MCM) is an arithmetic operation that multiplies a set of fixed-point constants {C1,C2,…,CM-1} with the same fixed-point variable X as shown in figure 2. From a circuit point of view, MCM dominates the complexity of the whole category of Linear Time Invariant (LTI) systems, such as, FIR/IIR filters, DSP transforms (DCT, DFT, Walsh, …), LTI controllers, crypto-systems, etc.

To be efficiently implemented, MCM must avoid costly multipliers. The hardware alternative must be multiplier less, i.e., using only additions, subtractions, and shifts. Therefore, the MCM problem is defined as the process of finding the minimum number of addition/subtraction operations.

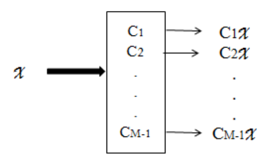

Figure 2. Concept of Multiple Constant Multiplier

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The computational complexity of MCM is conjectured to be NP-hard. Because of the increasing demand in high-speed and low-power design, MCM problem has been the focus of important researches during these last three decades. As a result, a big number of MCM algorithms have been proposed, mostly based on the acyclic directed graphs, or common sub-expression elimination, or the combination of both together.

The Finite impulse response filter is an important component for designing an efficient digital signal processing system. In this paper FIR filters are constructed, which consumes less power and area. Adders are the main building block for the construction of FIR filter. Addition is one of the fundamental arithmetic operations, used extensively in many VLSI systems such as microprocessors and application specific DSP architectures. The complexity of FIR filter is dominated by the number of adders and subtractors which are used to implement the co-efficient multipliers. To get optimized solutions, the original co-efficient is multiplied by constant value. This can be done by using multiplier block called Multiple Constant Multiplication (MCM) followed by accumulation of all products.

E. Processing of the Audio Biological System

In this section, the circuit level techniques that are used to implement the proposed algorithm into a power efficient hardware. Our approach is to detect the symptomatic patterns using acoustic non speech human signals with an increase in the efficiency of mathematical metrics, especially the Mel-frequency cepstral coefficients (MFCC) as shown in figure 3.

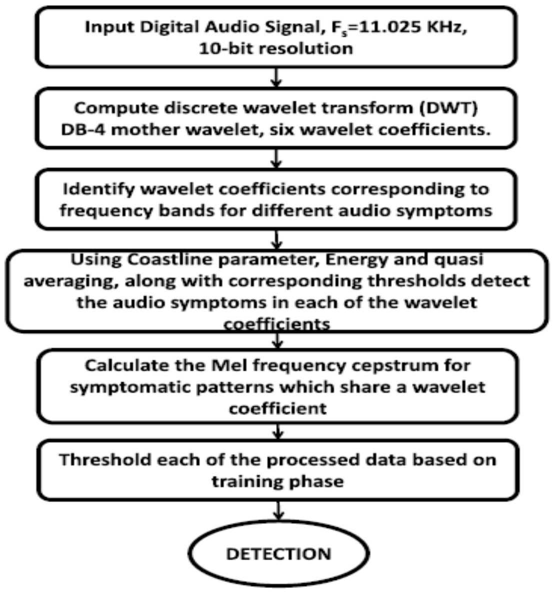

Figure 3. Proposed Methodology

F. Obtaining Discrete Wavelet Coefficients

The input data is the human audio recording of various symptomatic patterns, such as cough, sneeze, belch, wheeze and vomit. In this paper the input signal will be generated from audio file, such as MP3, wave, avi and so on. Using MATLAB to Convert the audio file to Hex Conversion, and Transfer the data to UART Communication at the baud rate of 115200, then stored the data to Memory as per signal with sampled digital format, the MATLAB GUI is as shown in Figure 4. This digitized sampled signal is streamed at the input of the algorithm at its sampling frequency (11.025 KHz).

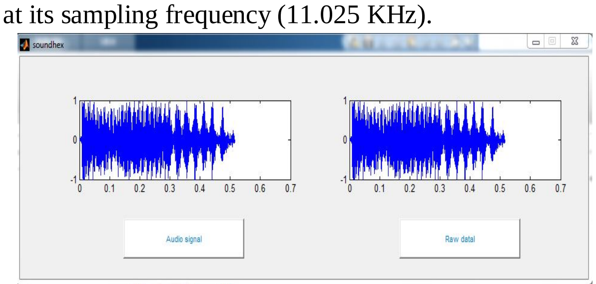

Figure 4. Streaming of Input Signal

The input has to be spectrally resolved using the DWT. Its multi-resolution ability to retain both temporal and spectral information justifies it to be the ideal choice for spectral resolution as compared with FFT or STFT. The DWT resolves the symptomatic patterns into narrow frequency bands or wavelet coefficients (Di) as shown in below Table 1.

Table 1:Mapping Of Dwt Coefficients To Frequency

DWT COEFFICIENT	RESOLVED FREQUENCY
D1	2.75-5.5kHz
D2	1375 Hz-2.75 kHz
D3	687 Hz-1375 Hz
D4	343 Hz-687Hz
D5	172.5 Hz-343 Hz
D6	86.25 Hz-172.5 Hz

The Daubechies fourth-order wavelet is used as the wavelet function for computation of the wavelet transform due to its optimal coarseness and smoothness to truly represent the signals of interest. The order of the selected mother wavelet is an algorithmic design decision, which has a direct impact on the complexity of its FPGA implementation. The various values of Di are classified as the coefficients of interest for specific symptomatic patterns. For instance, the acoustic patterns corresponding to wheezing and vomiting are resolved in the D5 and D6 wavelet coefficients, respectively. The pattern consistent with burp/belching is found in multiple coefficients (D4 and D5). The cough and sneeze signals have a common frequency spectrum and are resolved into a signal coefficients (D3). Another algorithm level design decision is the approximation of the filter coefficients used in computation of DWT. This has a negligible change in their frequency response.

Subsequent to the signal decomposition, the spectral as well as the temporal information of the signal is available for further processing. Although the symptomatic patterns are frequency resolved into separate wavelet coefficients, there are several sporadic spikes in the wavelet processed data, which might trigger false detection. Some of the coefficients are consisting of multiple symptoms too, while other patterns are resolved into multiple coefficients. In order to separate out these patterns further and reduce the noisy spikes to avoid false detection, these coefficients are subjected to various mathematical metric-based computation and MFCC base computation depending on the type of pattern to be detected.

G. Mel Cepstrum-Based Analysis

In the wavelet processed signal, it is observed that the cough and sneeze signals are resolved into the same DWT coefficient (D3). This is because these two symptomatic patterns have very similar frequency response standard mathematical metrics described previously, such as energy, CL, and so on, are not suitable for the efficacious classification of these signals. These signals are distinguished on the basic of the shape of the vocal tract while emitting them. Such distinguish ability is achieved base on MFCC computation.

The spectral envelope, which encodes the vocal tract shape in them, extracted by the MFCC algorithm. since our coefficient of interest is D3, it is only necessary to have the mel filters that overlap in the frequency band corresponding to D3 (689-1378 Hz), as shown in Figure 5. This results in the reduced number of mel filters.

The spectral energies of these filters are computed over a predetermined window size. The energies are passed on to a DCT block that de-correlates them producing the modified cepstrum coefficient. The first output coefficient of DCT corresponds to the dc component and can be ignored. The second and third coefficients correspond to the cough and sneeze patterns, respectively.

These MFCC-based parameters are then compared with a threshold, predetermined during training to detect the corresponding pattern. Depending on the type of signal being used, it is possible to modify the number of the mel filters in the filter bank and correspondingly modify the hardware implementation to achieve scalability and patient specific programmability.

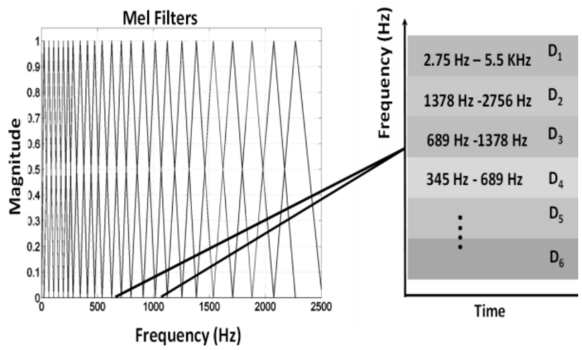

Figure 5. DWT Coefficient Mapping to Mel Filters.

H. Detection of Threshold

The threshold block consists of registers that are loaded with the prefixed threshold values corresponding to each individual acoustic pattern to be detected. These threshold values are fixed in training phase. Comparators in the threshold blocks are used to compare and raise the detection flag for each of the symptomatic pattern detected. The clock circuitry is used to synchronize all the operations in the system. The input data are streamed in at 11.025KHz. Each successive coefficients of the wavelet transform is computed at half the frequency as that of the previous coefficient. In this chapter the existing binary common sub-expression multiplier technique is discussed in detail and compared with the multiple constant multiplier technique. The merits of multiple constant multiplier technique is utilized in the realization of the proposed audio biological system with processing of each block in the detection of the audio biological signals. Thus the signal patterns can be detected accurately by utilizing the process stated in this section.

IV. RESULT AND DISCUSSION

Detection of the Audio Biological Signal

A. Synthesis Report

The synthesis report for audio biological system as shown in figure 6 infers the number of logic used, number of slice registers and slice LUTs which are involved in determining the area utilized and shows the power report the audio biological system which utilizes 2066mW..

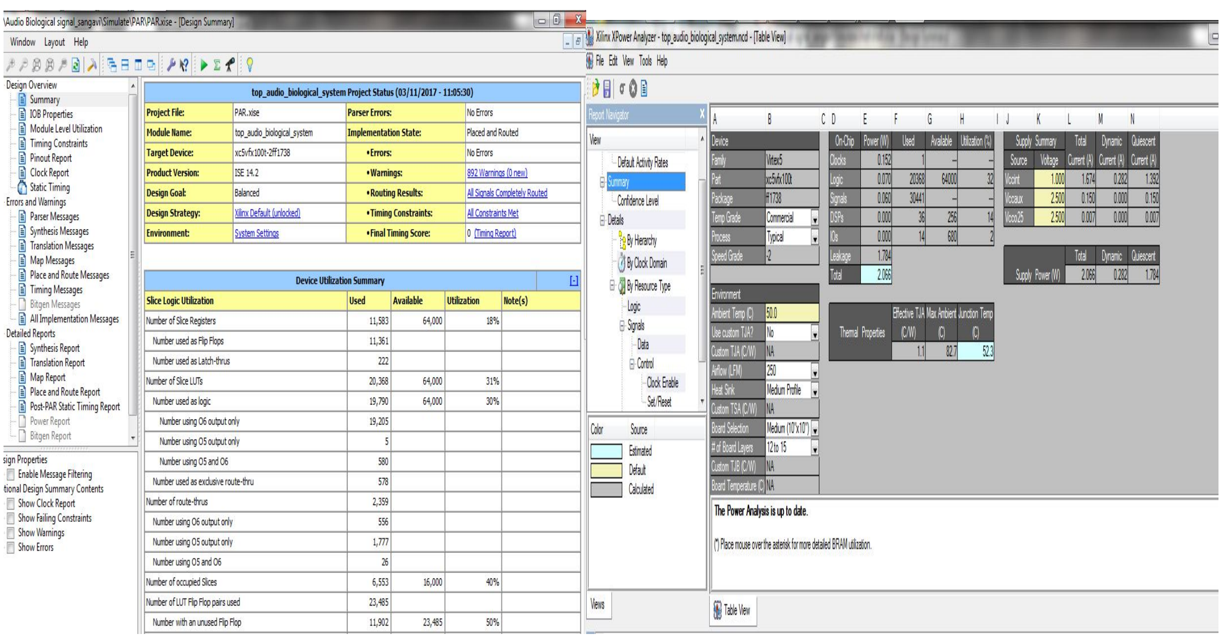

Figure 6. Synthesis and Power Report for Audio Biological System

In this section the comparison for the existing and proposed technique is shown in figure 21. this provides the efficient utilization of the MCM technique compared with the BSCE technique.

Table 2. Comparison Table for Existing and Proposed Technique

Design Metrics		BINARY COMMON SUB-EXPRESSION ELIMINATION TECHNIQUE	MULTIPLE CONSTANT MULTIPLIER TECHNIQUE	EFFICIENCY OF THE PROPOSED WORK (%)
AREA	SREG	352	205	41
	SLUT	588	409	30
POWER(mW)		114	14	87
DELAY(ns)		1.328	3.593	-

In this session simulation is performed for binary common sub-expression multiplier technique and the multiple constant multiplier technique. From the results it is inferred that the multiple constant multiplier using Radix-2r outperforms the binary common sub-expression multiplier in terms of hardware resources and power consumption. Finally the proposed multiplier technique outperforms the existing work in terms of area and power by 41% and 87% respectively. However this merit has been achieved at the cost of speed.

V. CONCLUSION

The proposed multiple constant multiplication algorithm is efficient in terms of area and power compared to the existing binary common sub-expression elimination technique. Experimental results demonstrate that by using multiple constant multiplier technique for detecting the audio biological signals the proposed architecture is able to interpolate the available area and power more efficiently compared to the existing binary common sub-expression elimination technique. In addition, a hardware sharing technique was used to reduce the hardware cost of the biological system.

The experiments are performed on the recorded audio signals observed from the patients by using MATLAB tool to determine the hexa-decimal value for each audio signal and the performance of the multiple constant multiplier by using Xilinx ISE tool. Experimental results revealed that the proposed multiple constant multiplier technique is superior to the existing binary common sub-expression elimination technique in terms of area and power.

Compared to the existing, the proposed system has improved performance with the area reduced along with the power of 2066mw. Hence the proposed system is power and area efficient.

A. Future Work

The future work would involve the improvement of the proposed algorithm for the signal processing and also to decrease the power and area consumption of the proposed multiplier architecture. The proposed system is tested with only few recorded audio signals. It may be extended to more audio signals. Algorithms that perform well but computationally complex can be implemented in VLSI so that they can be used in off-line application where processing speed never matters. The main research problem is area and low power architecture for the signal processing applications. Multiplier is frequently required in signal processing. Multipliers provide a high speed method for multiplication, but require large area for VLSI Implementations. So efficient multiplier architecture can be used to minimize the area and power in VLSI design.

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