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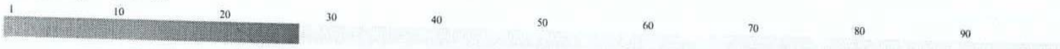

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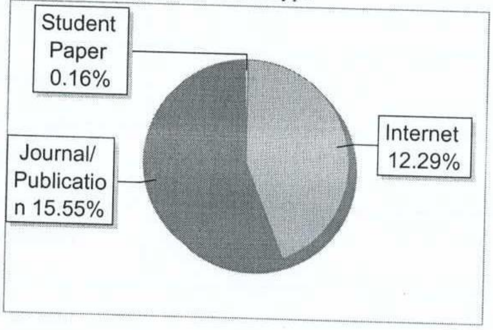

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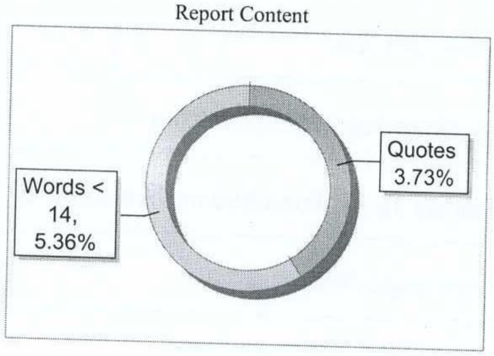

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HARDWARE-ORIENTED BLOCK MATCHING WITH SINGLE PRECISION FLOATING POINT DIVISION NEWTON-RAPHSON ALGORITHM FOR FAST MOTION ESTIMATION IN WIRE LESS NETWORKS

Abstract: *In the past few decades, we have been running towards the Digital Age. Telephony networks, fiber optic networks, Internet, satellites, third-generation (3G) wireless networks, these advanced transmission technologies and network infrastructure ensure digital information can flood into every corner around us. At the same time, the traditional media format has been gradually replaced by today's digital multimedia formats. These new formats and technologies shorten the distance between people, and improve the quality of lives. In this paper we have presented a Fast Motion Estimation technique known as Quadrant based search algorithm with parallel pipeline divider (QSPD) is constructed using Newton-Raphson Algorithm. Motion vector of the video sequence is estimated with Stationary Block Prediction (SBP). Stationary Block Prediction technique is used to find the block with zero motion. Implementation results of our proposed algorithm are compared with previous methods such as ARPS and Diamond search algorithm. PSNR value and motion estimation time of our proposed algorithm are better than that of previous method. In my research work, we concentrate on extending the block matching algorithms (BMA) from single reference frame to multiple reference frames. BMAs are selected due to their successes in the past video coding standards and the simplicity of regular data structure that is favorable for both hardware and software implementations. Analysis of motion vector distribution in multiple reference frames. Analysis the searching pattern of some out-standing BMAs and Developed a novel searching strategy for multiple reference frames based on the analyzed results. A novel approach to develop the fast motion estimation algorithm results are analyzed in Xilinx ISE and implemented in Spartan low power FPGA-6.*

Key Words: BMA, Peak Signal to Noise Ratio,. Big Cross Search Pattern, Small Diamond Search Algorithm.

I. INTRODUCTION

For a decade, various divider concepts are used in video coding techniques have always been advancing our multimedia technology. From VCD, DVD, Internet streaming, to video conferencing, all these multimedia applications indicate a break through of video coding techniques. A wide range of standards has been developed for different multimedia applications, such as ITU-T H.261 [1], H.263 [2], ISO/IEC MPEG-1 [3], MPEG-2 [4] and MPEG-4 [5]. These standards focus on different application profiles in terms of picture quality, bit-rate, latency, network capability and complexity. They provide a common area for the interested bodies to work on and open a new market for consumer electronics. The new technology also revolutionizes the forms of different fields from time to time, for example, entertainments, teaching, medicine, geography, and meteorology.

Although today network bandwidth and storage continuously increasing, the endless demands of better image quality and faster communication can fill up the room easily. Supposing a digital video of 24-bit color depth with 720x480 resolutions is transmitted at 30 frames per second (fps), and then it requires 248Mbps bandwidth. High definition TV (HDTV), video-on-demand (VOD), video email and video conferencing and telephony are now the hottest and most demanded multimedia services. However, these services also require huge data bandwidth to provide the high quality pictures, and could easily consume a great portion of the network bandwidth of a service provider. Consequently, the providers would need to invest heavily on the network infrastructure. This is also the main concern that deters them from trying this business. As a result, many academic and industrial researchers are working on a more efficient video coding standard with different divider techniques integrated on it which is suitable for video applications in the limited bandwidth environment.

1.1 New Emerging Standard: H.264/MPEG-4 AVC

H.264/MPEG-4 AVC [6] is the latest video coding standard developed by the Joint Video Team (JVT) which is formed with ITU-T Video Coding Experts Group (VCEG) and ISO/IEC Motion Picture Experts Group (MPEG) in 2001. VCEG was formed in 1997. It is chartered as Question 6 of Study Group 16 to establish international standards for conversational and non-conversational audio/visual applications. MPEG was formed in 1988. It is chartered as Work Group 11 of Subcommittee 29 of JTC1 to develop moving pictures coding standards for a wide variety of digital video applications like DVD (storage media) and VOD (broadcasting).

There are 3 reasons for the need of a new industry standard:

- The cost for processing power and memory has reduced. More heavy coding strategies can be supported;
- More network supports are available for coded video data;

- Video coding technology has been advanced. The previous codec is not up to date.

JVT gathers the preeminent experts from these 2 leading international standards organizations to develop a new standard that meet the ever-growing need for higher compression of various video applications like digital storage media, TV broadcasting, video conferencing and internet streaming. The standard will eventually be known as ITU-T H.264 and ISO/IEC MPEG-4 Part 10 Advanced Video Coding (AVC). It is supposed to be published in the mid 2003 [7].

This new standard significantly out-performed the existing video coding standards. The previous best video codec, H.263 and MPEG-4 Visual (Part 2), based on the video coding technology of around 1995 are mainly used for low bit-rate communication and multimedia streaming on web. MPEG-2 is the video coding format for HDTV and DVD. H.264 can save half of the bit-rate when compared with the former, and only use about quarter of that for the latter [8]. In other words, we can have 2X or 4X the video quality by renewing the current video codec while keeping the same bandwidth requirements. A recent research [9] on the average US household consumption of digital broadcast TV suggests that even a little improvement on the video compression efficiency (e.g. 10%), would reduce 20X of current Internet backbone traffic. This means that the new standard not only pursues higher quality, but also opens new opportunities for various bandwidth demanding video applications, especially for those mobile devices that have only limited bandwidth in the wireless network such as General Packet Radio Service (GPRS) and 3G.

1.2 Statement of the Problem

The significant gain of compression efficiency of H.264 is at the expense of increased computation and complexity. For examples, 4x4 DCT transform, advanced intra/inter-prediction modes, tree-structured macroblock partitioning, quarter-pixel motion compensation and multiple reference frames, all these features help to increase the compression efficiency, but introduce huge amount of computation and complexity requirements of hardware [6].

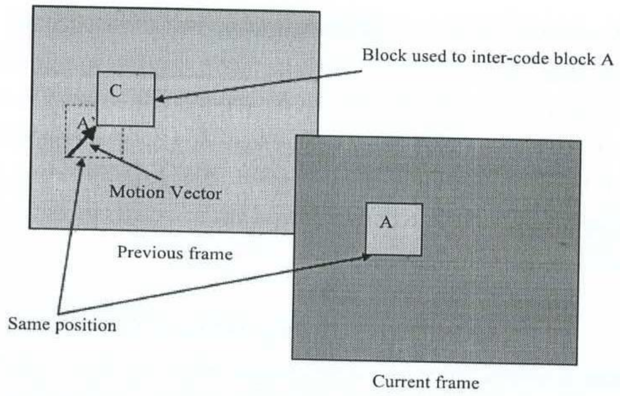


Figure 1.1: Motion compensation

In a hybrid-coding [10] video encoder, most of the computation is spent on motion estimation. Hybrid-coding is virtually the basis of all video coding standards, in which intra-frame coding and inter-frame coding are combined to use to reduce the spatial and temporal redundancy. Motion estimation (ME) and motion compensation (MC) are the two major components of inter-frame coding for video compression. As depicted in Figure 1.1, motion compensation is a kind of predictive coding that explores the data correlation in temporal domain by compensating for the displacement of objects between the reference frame and current frame. On the other hand, motion estimation is the process to find out the motion vector (MV) which represents the displacement of the objects. A good prediction can substantially increase the compression efficiency. Nevertheless, such kind of motion-compensated prediction technique is computationally intensive. Up to 80% computational power of an encoder can be consumed by motion estimation [11], and this is the case for exhaustive searching of all candidate blocks in a single reference frame only. In H.264, motion estimation is allowed to search on multiple reference frames with multiple partition sizes at quarter-pixel accuracy, and its reference software adopts full search scheme. Certainly, the computational load due to motion estimation must be increased greatly and directly proportional to the number of reference frames. VideoLocus, one of the leading developers of video compression technology, has recently developed a video evaluation platform for H.264. It shows that, without hardware acceleration, a 2GHz P4 computer requires 10 seconds to encode a single frame for H.264 [12]. This is absolutely a challenging requirement for implementing a software encoder with today computational power, especially for mobile computing devices. Obviously, it indicates there is an eager need for faster motion estimation strategy. In all the above mentioned optimization is done with respect to algorithmic level where as we are concentrating on reconfigurable design through optimizing divider and by introducing parallel pipelining architecture.

In Section 2, an overview of the H.264 coding standard will be introduced. We will explain the impact of the advanced prediction features on the compression efficiency of the new codec. In Section 3, a summary of some well-know motion estimation algorithms and two recently proposed multiple-frame-selection algorithms will be presented with divider approach. In Section 4, a wholesale analysis of motion vectors distribution on multiple reference frames of various sequences will be shown with divider approach based on Newton raphston method. Based on the result, two novel fast motion estimation algorithms for multiple reference frames are proposed and the possible outcome will be discussed. Finally, our research will be concluded in Section 5.

2. EXISTING METHOD OVERVIEW OF H.264

H.264 is the most up-to-date video coding technology. Many advanced features are assembled in the standard. These features include 4x4 DCT transform which replaces the 8x8 DCT transform in previous standards; advanced intra-prediction which has up to 13 prediction modes for luminance blocks; advanced inter-prediction which support motion compensation in multiple reference frame with multiple macroblock partition size at quarter-pixel accuracy. In this chapter, we mainly focus on discussing about the prediction part of the new codec and how it essentially increases the video compression efficiency.

2.1 H.264 Codec Design

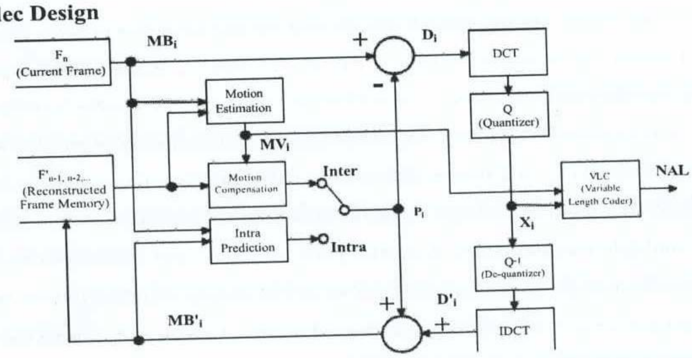


Figure 2.1: H.264 Video Encoder

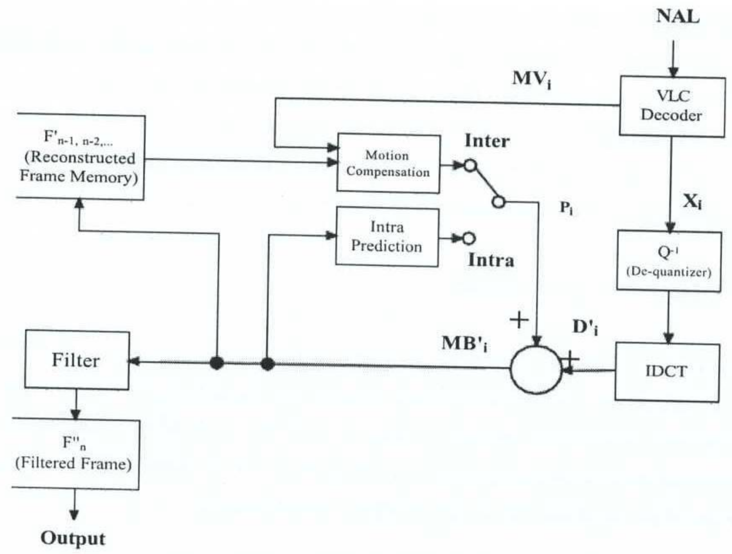


Figure 2.2: H.264 Video Decoder

Similar to other standards, the H.264 codec (encoder and decoder pair) consists of some basic functional elements like transformer, quantizer, variable length coder, and their counterparts. The encoder is shown in Figure 2.1. The encoding path starts from left to right. A raw frame F_n is input to the encoder. Each frame is partitioned into blocks with 16x16 pixels that they are called macroblocks. The encoder processes the macroblocks one by one. A macroblock MB_i is encoded either in intra or inter mode. For both modes, a prediction P will be produced. In inter mode, P_i is formed by motion compensation of the reference frames, $F_{n-1, n-2, \ldots}$, stored in frame memory. The motion estimation finds the best match of the current block from one or more reference frames and generates the motion vectors MV_i which represent the displacement of the block. In intra mode, P_i is predicted from the previous reconstructed neighboring macroblocks. The prediction is then subtracted from MB_i to produce the residual data D_i. The residual data is transformed and quantized for lossy compression to give X_i which contains the quantized DCT coefficients. These coefficients and motion vectors are entropy encoded to form a bitstream with other side information like quantization factor and prediction mode. It is sent to the Network Abstraction Layer (NAL) for storage and transmission. In return path, X_i is de-quantized and de-transformed. As quantization is a lossy process, the reconstructed residual data D'_i is not exactly the same as original. It is then added to P_i to form a reconstructed macroblock, and stored in the frame memory for further reference.

In the decoder shown in Figure 2.2, it is quite similar to the reconstructing path of the encoder. The decoding path starts from right to left and up to down. The data received from NAL are processed inversely as in the encoder. Eventually, the reconstructed frame is filtered to alleviate the artifacts due to lossy compression. One should notice the codec design that both the encoder and decoder use the identical references for prediction. Otherwise, error will accumulate on successive decoded frames. That is also why we have a reconstructing path in the encoder. Newton-Raphson computational division algorithm with parallel pipelining approach is introduce in the proposed and existing method in order to optimize the area overhead and delay with less power utilization as discussed below.

2.2 Intra Prediction of Macroblocks

The intra coding of H.264 is quite different to that of the previous standards. In H.264, if a macroblock is coded in intra mode, a prediction block is always generated from the neighboring decoded samples in spatial domain. Whereas, in MPEG-4 Visual and H.263+, only some of the transformed coefficients can be predicted from neighboring samples in frequency domain, and there is even no prediction in intra mode for the older standards like H.261, MPEG-1 and MPEG-2.

There are three types of intra coding supported in the standard, two for luma blocks and another one for chroma blocks. No matter which type of macroblocks, intra prediction is not allowed performing across slice boundaries to keep each slice independent of decoding.

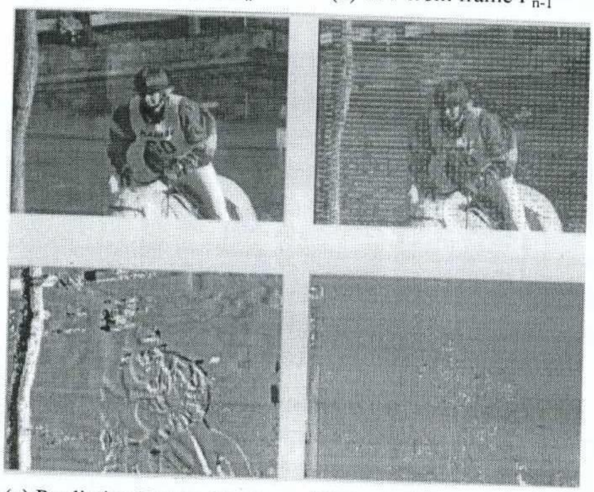
2.2.1 Intra_4x4 prediction for luma samples

Q	A	B	C	D	E	F	G	H
I	a	b	c	d				
J	e	f	g	h				
K	i	j	k	l				
L	m	n	o	p				

Figure 2.3: Label of samples used for intra prediction

3. MATERIALS A AND METHODS OF PROPOSED AND CONVENTIONAL FAST MOTION ESTIMATION ALGORITHMS

(a) Original raw frame F_n (b) MV from frame F_{n-1}



(c) Prediction Error without using MC (i.e. all MVs are 0) (d) Prediction Error if MC is applied

Figure 3.1: Prediction with / without motion compensation

Motion estimation plays an important role in video compression. In general, it can improve the compression efficiency by utilizing the temporal redundancy between adjacent pictures. As depicted in Figure 3.1, the motion compensated picture 3.1(d) has much less error than the one without it 3.1(c). It means that the necessary coded information is reduced. In H.264, as well as all other video coding standards, only the syntax elements of motion estimation is specified while the algorithm is left open. Researchers and vendors can have their room for development and competition. As a result, many kinds of motion estimation algorithms have been proposed [10], [14], [15]. Among those algorithms, one that commonly adopted by different communities and standards as a reference is the block matching algorithm with Newton-Raphson computational division algorithm.

3.1 Block Matching Motion Estimation

Most of the block matching motion estimation (BMME) algorithms rely on the following assumptions:

- The uniform illumination is along the motion axis;
- The uncovering background problem is not present;
- The pixels inside a block have the same shift between successive frames.

The first assumption states the problem that illumination may introduce different optical flows that are not related to the motion. The second assumption addresses the problem of scene change and uncovered objects that they are not exist in the previous frame. The last assumption points out that different motions may present within a block.

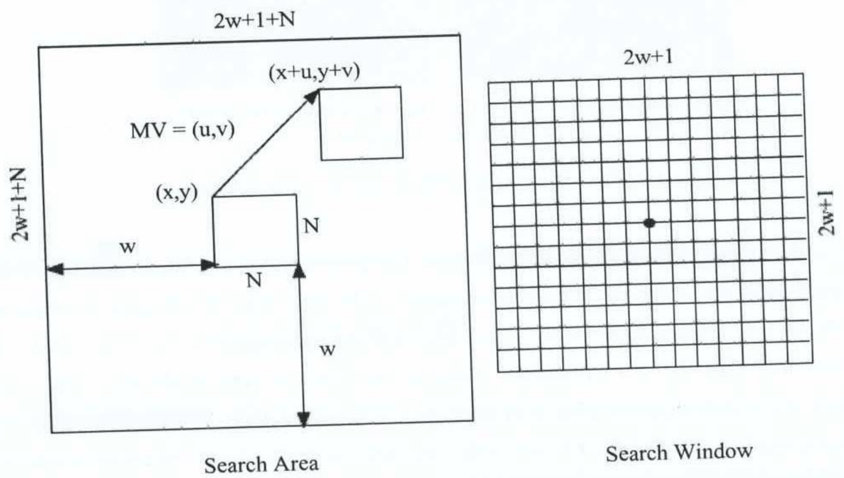


Figure 3.2: Search area and search window for block matching

A block matching algorithm finds the best match of the current block from the search area of the reference frame. Each frame is segmented into blocks of 16x16 pixels (macroblock) for processing.

These blocks are given a search window where they are located in the center. Figure 3.2 shows a general case. Supposing the window size is w, then there is total $(2w+1)^2$ candidate blocks to be searched. The actual searched area on the reference frame should be to $(2w+1+N)^2$ if the block has a size of NxN. The best match is usually evaluated by a cost function which based on the block distortion measure (BDM) (e.g. Mean Square Error (MSE), Mean Absolute Error (MAE) and Sum of Absolute Difference (SAD)) or other criteria like bit-rate and number of motion vectors. The motion vector (MV) is then defined as the displacement of the best match from the current block. In all the reference models of the standards, full search (FS) scheme is adopted. In this case, we need to search through all the candidate blocks as mentioned above. The exhaustive FS can typically have 60% to 80% [11] computation of an encoder. Thus, fast searching algorithms are desirable to speed up the searching time and reduce the loading by optimizing divider approach.

3.2. Overview of our conventional hardware architecture

Figure.3.1(a) shows the hardware architecture of the conventional Quadrant based search algorithm with parallel pipelining divider and stationary motion prediction (SBP). It consists of current data buffer, reference data buffer, process element array, SAD comparator and also SBP block. From the figure the current data buffer and reference data buffer used to store the pixels values of current blocks and reference blocks. Processing element (PE) array is used to calculate the SAD value between the current and reference data. SAD comparator used to find the minimum SAD value from the calculated SAD values. SBP block has predicted threshold value and it used to compare the SAD of the centre point in the QSPD. In this architecture they have used single pipelining approach for divider concept is used as shown in the Figure.3.1.(b) .

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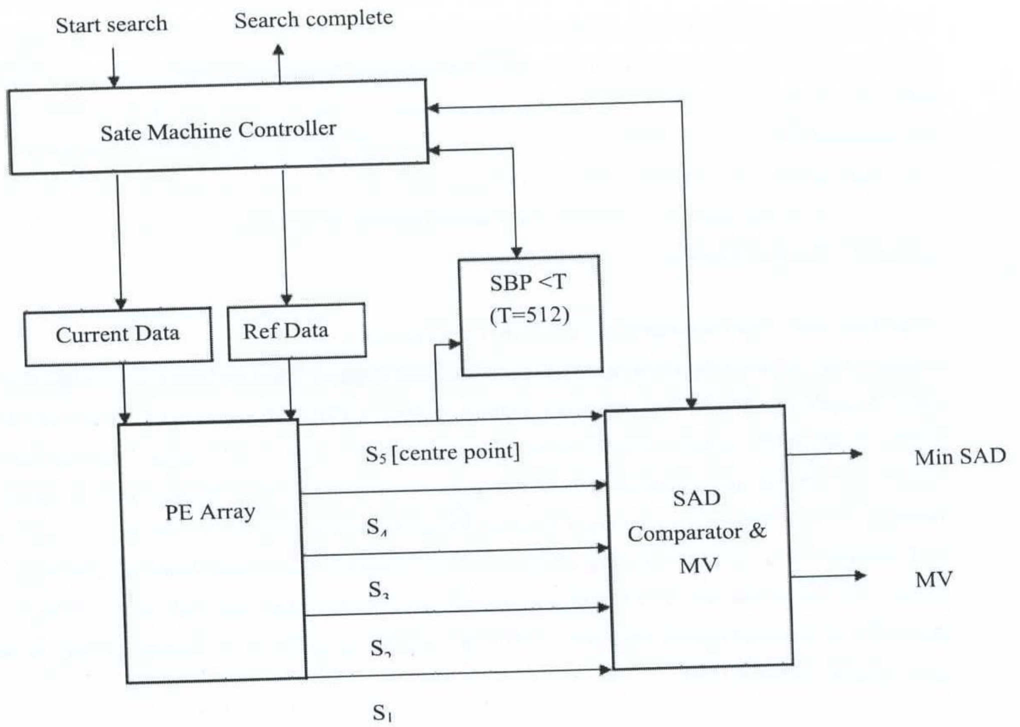


Figure.3.1. (a) Hardware architecture of Conventional method

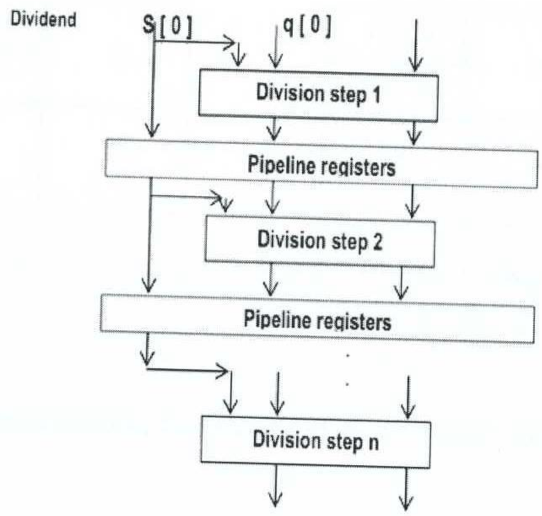
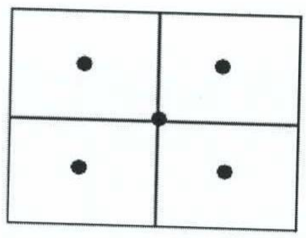
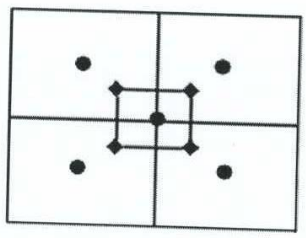


Figure.3.1. (b) Conventional single pipelining Method Flow diagram

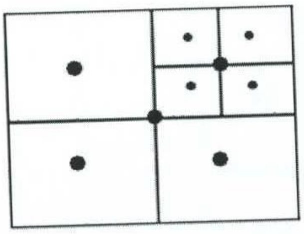
In order to further speed up the search, Stationary Block Prediction is introduced, which is also known as Zero Motion Prejudgment. For zero motion blocks, the block distortion between the current block and reference block is very low as compared to moving blocks. Therefore to find a stationary block first step is to find the block distortion for current block with the collocated block in the reference frame. Second step is to compare it with a predetermined threshold. The current block is declared as a stationary block if the block distortion is below a predicted threshold (T). The block is assigning motion vector (0, 0) and search is terminated. The success and accuracy of this prediction depends upon the accuracy in prediction of T. In [14] author suggested a fixed T value as 512. As shown in the figure.3.2.



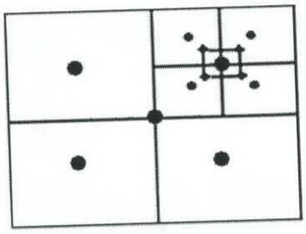
a)Quadratic search pattern (QSP) formation



b) Small Square Search Pattern (SSSP)



c) If any of one quadrant point has minimum SAD value, it will be taken as centre point for



d) The process continue until find the motion

Figure.3.2 (a, b, c & d): Quadrant search pattern algorithm

3.3 Fast BMA for Single Reference Frame

In this section, five fast BMAs will be briefly described. All of them apply the sub-sampling technique to reduce the number of search points. They commonly have an assumption that *the BDM decreases monotonically towards the global optimal point* [16]. Therefore their searching strategies gradually converge in the direction that the minimum BDM is detected. Although this assumption does not always hold for the real world sequences, low BDM sub-optimal points can still be found if suitable strategies are used to prevent being trapped by some local minima.

3.2.1 Three-Step Search Algorithm (3SS)

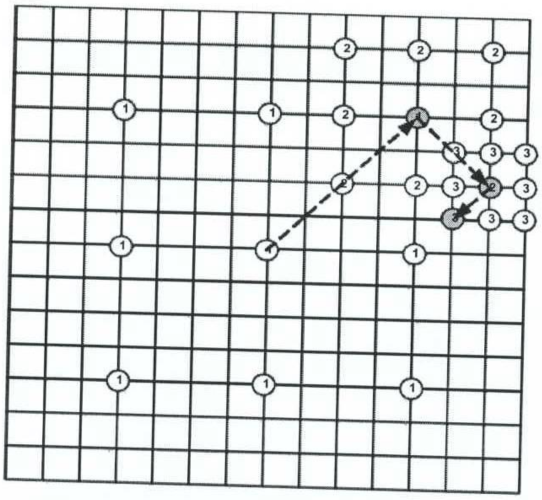


Figure 3.3: Example of 3SS

3SS was proposed by T. Koga in 1981 [17]. It applies a coarse-to fine approach that assumes the motion vectors of video sequences are uniformly distributed. As shown in Figure 3.3, the search starts from the center of the search window. It uses a nine searching point's grid for each step. The initial step size is equal to $\lceil w/2 \rceil$ where w is the window size and $\lceil \ \rceil$ is a round up function. Each successive step has half the size of the last step. The minimum BDM point is defined as the center of the next step. The search will be terminated when the step size is equal to one. Proposed algorithmic flow is shown in the figure.3.4.Further the algorithm is implemented in FPGA as shown in the figure.4.5,4.6 and 4.7

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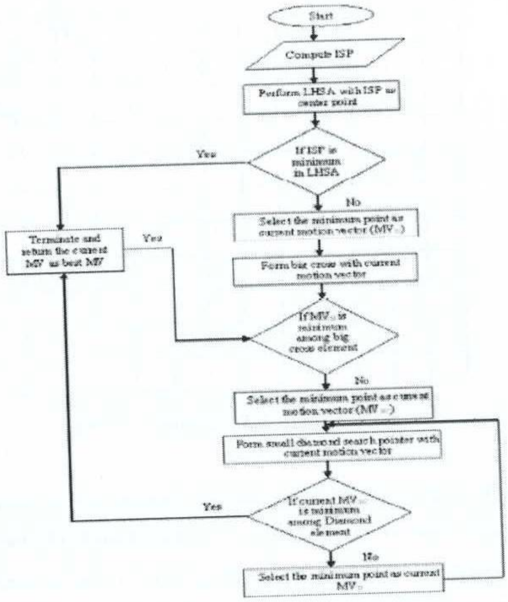
Figure 3.4.Proposed Algorithmic flow.



Algorithm Flow:

Step 1: Determine the ISP centre point from the reference frame and provided to searching window of current frame block.

Step 2: find the SAD1 value for current frame by using Large Hexagon Search Pattern (LHSP) and compare with ISP value.

Step 3: If the SAD1 of the centre point in the QSPD is less than the ISP, diminished SAD1 value is considered to be the last value or else determine the minimum SAD value and set the centre point as ISP2 value.

Step 4: find the SAD2 value for current frame by using Big Cross Search Pattern (BCSP) and compare with ISP2 value.

Step 5: If the SAD2 of the centre point in the QSPD is less than the ISP2, diminished SAD2 value is considered to be the last value or else determine the minimum SAD value and set the centre point as ISP3 value.

Step 6: find the SAD3 value for current frame by using Small Diamond Search Algorithm (SDSA) and compare with ISP3 value.

Step 7: If the SAD3 of the centre point in the QSPD is less than the ISP3, diminished SAD3 value is considered to be the last value or else determine the minimum SAD value and set the centre point as ISP4 value and repeated same procedure from step6 to step7 until to get efficient motion vector(MV)value .Further optimization in divider algorithm is shown below Figure.3.5. Newton-Raphson computational division algorithm with parallel pipelining approach is introduce in the proposed and existing method in order to optimize the area overhead and delay with less power utilization as discussed below.

A. Newton-Raphson computational division algorithm:

In paper describes a single precision floating point division based on Newton-Raphson computational division algorithm. The Newton-Raphson computational algorithm is implemented using 32-bit floating point multiplier and subtractor. The salient feature of this proposed design is that the module for computing mantissa in 32-floating point multiplier is designed using a 24-bit Vedic multiplication (Urdhva-triyakbhyam-sutra) technique. 32-bit floating point multiplier, designed using Vedic multiplication technique, yields a higher computational speed, hence, is efficiently used in floating point divider. Another important feature is the efficient use of device utilization parameters and reduced power consumption. An advantage of the Newton-Raphson algorithm is the higher versatility and precision. As shown in the Figure.2.3.

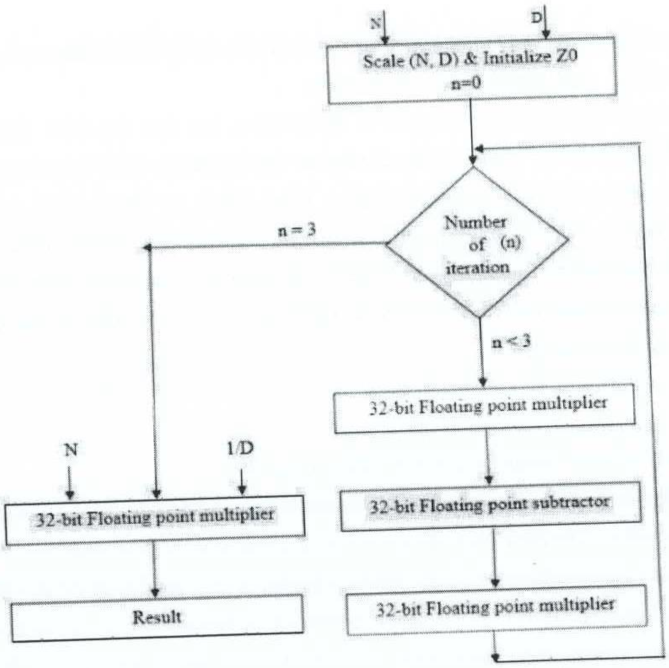


Figure.3.5.A Flowchart of the 32-bit floating point division using Newton-Raphson method.

4. RESULTS AND SIMULATION OF MULTI-FRAME MOTION VECTOR DISTRIBUTION

In this chapter, two novel fast BMAs are proposed for motion estimation of multiple reference frames. As the motion vector distributions give significant information about the real-world motion properties, which in turn affect the searching strategies in motion estimation. Some analyses on the motion vector distributions and rate distortion are conducted beforehand. Based on the results, the detailed steps of our methods are developed.[18,19]

4.1 Experiments and Analyses

Sequence	Format	Motion
Claire	CIF (352x288)	Small
Miss America	CIF (352x288)	Small
Sales	CIF (352x288)	Small

Garden	SIF (352x240)	Large
Football	SIF (352x240)	Large
Tennis	SIF (352x240)	Large

Table 4.1: Six testing video sequences

Three kinds of analyses are conducted on a series of experiments to find out the distribution of motion vectors and importance of number of reference frames. In these experiments, full search motion estimation is simulated on six CIF/SIF video sequences, Table 4.1. Three of them contain small motions, and the others contain large motions. Motion estimation is performed for 80 frames at fixed 16x16 block size with integer pixel accuracy. The search window size is set to w=7, w=15 and w=30. Ten sets of experiments are done for 1 to 10 allowed number of reference frames.

4.1.1 Distortion Gain

r / Sequences	MAE per frame with used no. of reference frames = r									
	1	2	3	4	5	6	7	8	9	10
Claire	105024	102651	101288	100747	100310	99791	99434	99197	98899	98705
missA	231422	221222	218201	216380	215231	214290	213610	213081	212691	212333
Sales	287478	153003	152434	150201	150026	148979	148865	148119	148066	147539
Garden	713569	689149	685035	682286	678514	675776	674721	674346	673926	673652
Football	581333	564103	557374	551346	549514	547516	546517	545114	544445	543332
Tennis	520248	492739	484518	479861	474579	472149	470124	468311	466570	465014
Avg MAE per pixel	4.607	4.229	4.183	4.149	4.124	4.106	4.096	4.087	4.080	4.072

Table 4.2: MAE per frame for each sequence with w=7

r / Sequences	MAE gain (%) per frame with used no. of reference frames = r									
	1	2	3	4	5	6	7	8	9	10
claire	N/A	2.26	3.56	4.07	4.49	4.98	5.32	5.55	5.83	6.02
missA	N/A	4.41	5.71	6.50	7.00	7.40	7.70	7.93	8.09	8.25
sales	N/A	46.78	46.98	47.75	47.81	48.18	48.22	48.48	48.49	48.68
garden	N/A	3.42	4.00	4.38	4.91	5.30	5.44	5.50	5.56	5.59

football	N/A	2.96	4.12	5.16	5.47	5.82	5.99	6.23	6.35	6.54
tennis	N/A	5.29	6.87	7.76	8.78	9.25	9.63	9.98	10.32	10.62
(i) Average	N/A	10.85	11.87	12.60	13.08	13.49	13.72	13.94	14.11	14.28
(ii) Unit of load	1.00	2.00	3.00	4.00	5.00	6.00	7.00	8.00	9.00	10.00
Efficiency = (i)/(ii)		5.43	3.96	3.15	2.62	2.25	1.96	1.74	1.57	1.43

Table 4.3: MAE gain in various no. of reference frames with w=7

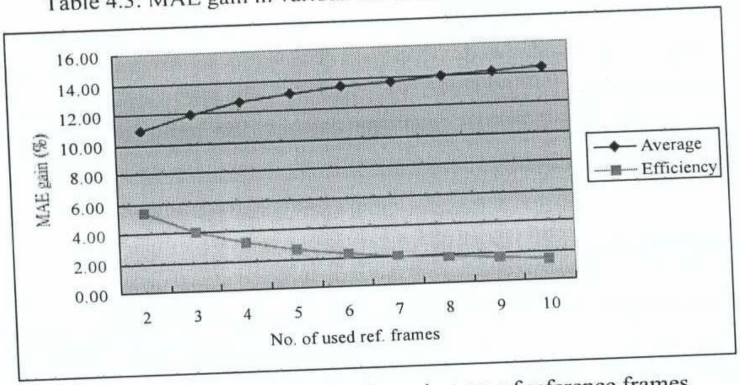


Figure 4.1: Change of MAE gain against no. of reference frames

Two curves are shown in Figure 4.1. The blue one is the average MAE gains of the six sequences. The MAE gain is defined as the relative reduction of MAE to the value counted using single reference frame. In order to represent the efficiency of increasing the number of reference frames, the MAE reduction is divided by the computational load of each frame can be optimized because of Newton-Raphson computational division algorithm with parallel pipelining approach because As the load increases linearly for increment of reference frames, the efficiency in fact decreases and tends to constant after 5 to 6 frames which can be optimized with this method. The efficiency is shown by the pink curve. The analysis reveals that the improvement is insignificant when searching more than about 5 reference frames. In contrast, the system would be burdened with the complexity.Implementaion on FPGA results are shown in figure 4.5 and 4.6.

4.1.2 Motion Vector Density

n / Sequences	MV distributions (%) in 10 ref. frames F(t-n)									
	1	2	3	4	5	6	7	8	9	10
Claire	64.00	8.00	9.00	4.00	3.00	4.00	2.00	2.00	2.00	1.00

missA	22.00	18.00	11.00	9.00	8.00	8.00	6.00	7.00	6.00	6.00
Sales	16.00	37.00	1.00	15.00	0.00	11.00	0.00	11.00	0.00	10.00
garden	77.00	12.00	3.00	2.00	2.00	2.00	1.00	0.00	0.00	0.00
football	53.00	14.00	5.00	12.00	2.00	4.00	1.00	5.00	1.00	4.00
tennis	61.00	19.00	6.00	2.00	4.00	2.00	2.00	1.00	1.00	2.00
Average	48.83	18.00	5.83	7.33	3.17	5.17	2.00	4.33	1.67	3.83
Accumulation	49	67	73	80	83	88	90	95	96	100

Table 4.4: Motion vector distribution in 10 reference frames

Table 4.5 shows that device utilization of our proposed approach is compared with that of previous method Full Search (FS) and Diamond search algorithm (DS). QSPD used 306 LUT's respectively than our proposed approach which uses 413 slices. Our proposed approach uses 239 slice flip flops and 186 LUTs. Maximum operating frequency of our proposed approach is 415.317MHz. Besides our proposed approach is good in total on chip power consumption as shown in the table 4.5 after integrated with parallel pipeline approach.

Attributes	FS[24]	DS[25]	QSPD	BMA
No.of slices	14.7K	7.8K	186	239
No.ofFlipFlops	-	11.3K	226	153
No.of 4 I/P LUTs	-	10.8K	306	413
Minimum period	-	-	2.408ns	12.205ns
Max frequency (MHz)	149.2	246.5	415.317MHz	81.933Mhz
Power(mw)	-	-	0.187	0.052
Number of PE	16	16	5	6
Search range	± 4 *pixels*	± 16 *pixels*	± 8 *pixels*	± 25 *pixels*

Table 4.5: Motion vector distribution in 10 reference frames

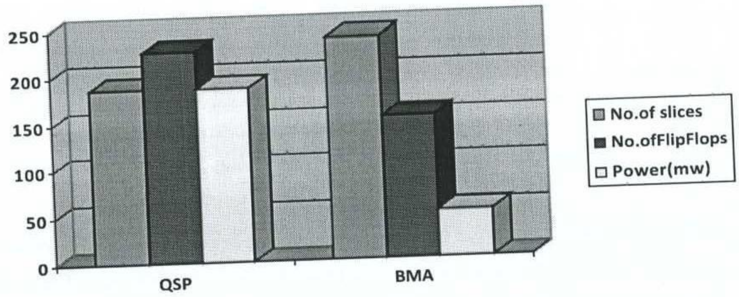


Figure.4.6 Optimization results of the existing and proposed approach.

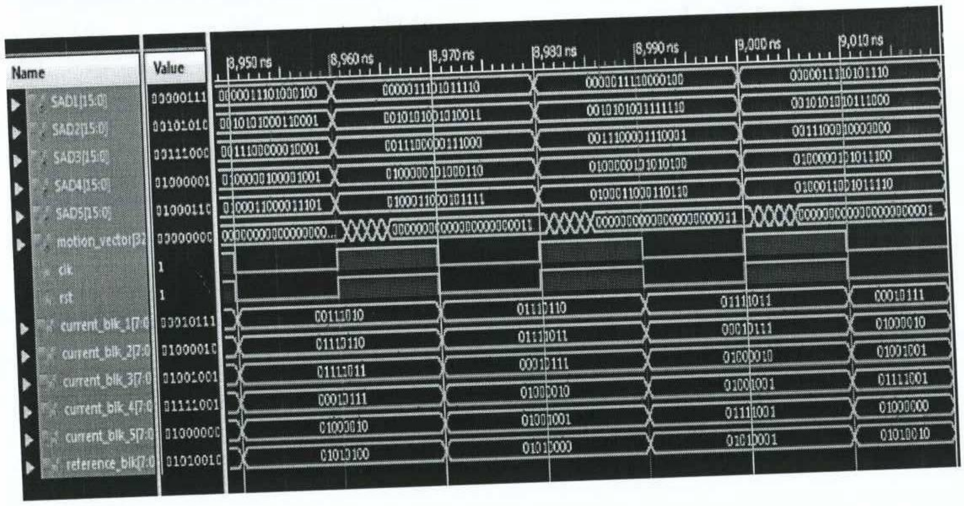


Figure 4.7: Simulation result of the conventional algorithm

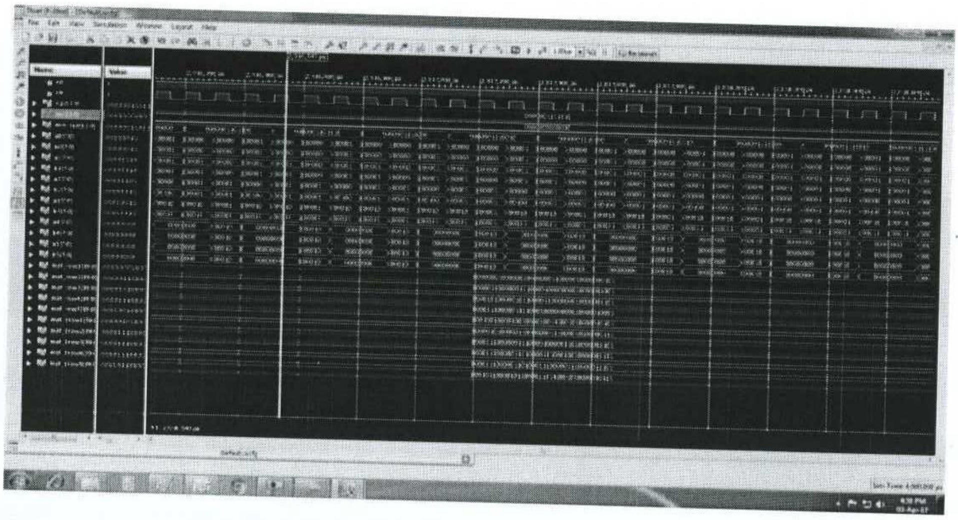


Figure 4.8: Simulation result of the proposed algorithm

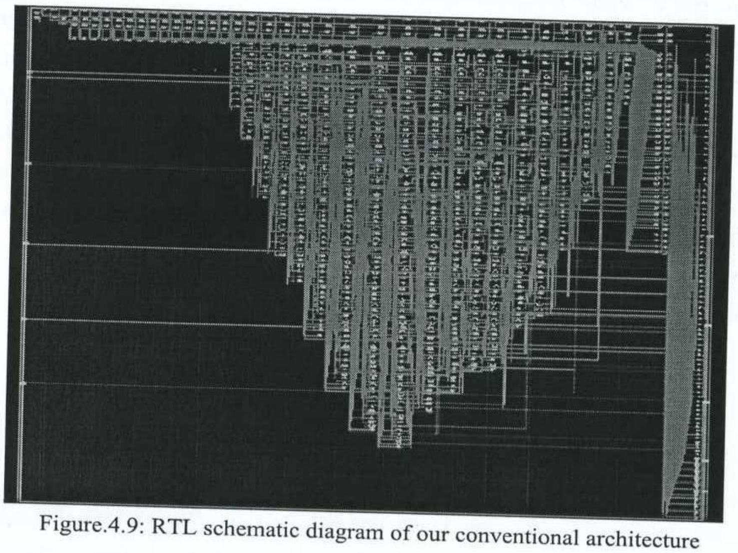


Figure.4.9: RTL schematic diagram of our conventional architecture

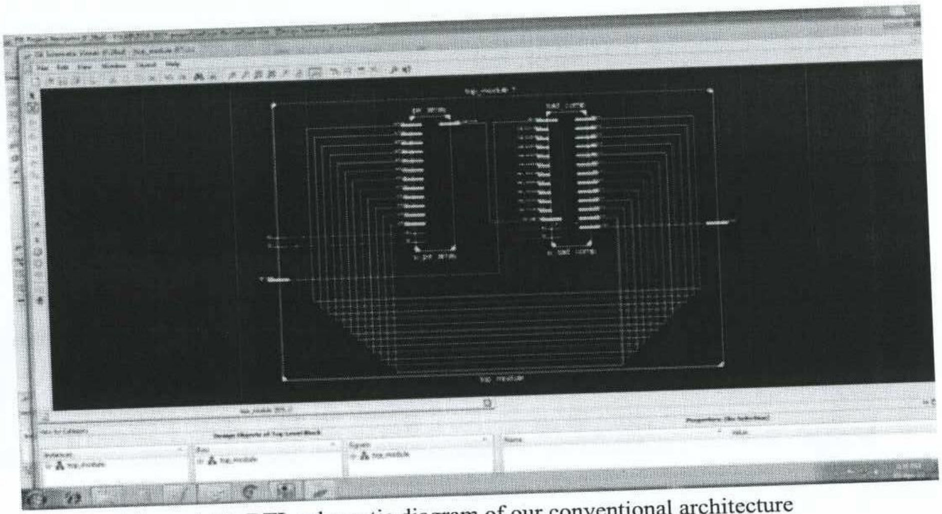


Figure.4.10: RTL schematic diagram of our conventional architecture

The power report produced by the Xilinx XPower Analyzer instrument appeared in Table.4.5. Our proposed architecture devours **0.052** total on-chip power Figure 4.7 ,4.8,4.9 and 4.10 demonstrates the simulation results and RTL (Register Transfer Logic) of diagram of our proposed architecture as well demonstrates the reproduction aftereffects of the foreseen engineering. In the above mentioned table (4.1 to 4.4), motion vectors of the information outlines indicated relating to the square address of the casing furthermore the clock flag. Motion vector has been found for the input video frames by comparing the number of current frames with one reference frame as shown in the above table comparison.(4.1 to 4.4).

5. CONCLUSION

In my research, video coding techniques and the advanced prediction features of H.264 are studied. With the multiple reference frames motion compensation, a wholesale motion vector distributions analysis is conducted in both spatial and temporal domains. The experimental results show that (1) cross-center-biased remains in multiple reference frames; (2) five reference frames is an optimal for complexity and prediction quality; (3) search window size does not affect motion vector distributions. Based on the characteristics we found, two novel 3D fast motion estimation algorithms are proposed to exploit the cross-center-biased behavior in real-world sequences with Newton-Raphson computational division algorithm with parallel pipelining approach. As the rationales are supported by the analysis results, it is supposed that our algorithms will have a significant gain in the speed while keeping similar PSNR quality to FS. Implementation results of our proposed algorithm are compared with previous methods such as conventional UM Hexagon algorithm and shows higher results due parallel pipelining approach in divider. In our proposed method, we consider the Initial Search Point (ISP) to determine the motion vector efficiently. Conventional UM Hexagon technique is used to find the block with zero motion. The efficiency of the proposed algorithm is evaluated in terms of Mean Square Error (MSE) and Peak Signal to Noise Ratio (PSNR). Our future work is the simulation in H.264 reference model.

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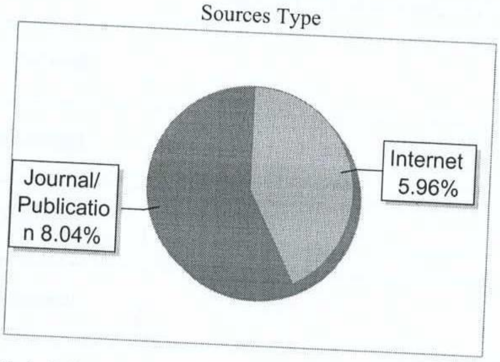

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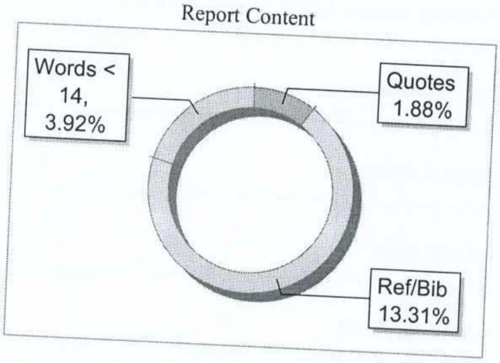

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RFID based Prepaid energy meter to measure the electricity usage of domestic applliances and to pay the electricity bill automatically

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Abstract:

The project's goal is to reduce the backlog at the electricity billing counters and automatically limit usage of electricity if the bill is not paid. Consumers may find out how many power units they are using and how much it costs by using RFID technology. Energy theft and power theft are the only current problems with the current invoicing system. The use of energy prepayment meters helps to solve this issue. Prepaid refers to the idea of "paying before use," and it's a similar idea to prepaid mobile. In this case, a microcontroller has been introduced together with a smart card used for recharge. When a consumer inserts a smart card into the reader, the card reader reads the information that has been placed on the card and deletes information so that the card cannot be used by another person. And the quantity of increased power consumption will reduce.

Keywords :

RFID, Micro controller, crystel oscillator,LIQUID CRYSTAL DISPLAY(LCD),

Introduction

A computer system called an embedded system is made to carry out one or a small number of specific tasks, frequently under real-time computing limitations. It is integrated as a component of a finished product, frequently with physical and mechanical components. A general-purpose computer, such as a personal computer (PC), on the other hand, is made to be versatile and to satisfy a variety of end-user needs.

Today's commonplace devices are controlled by embedded systems. One or more main processing cores—typically microcontrollers or digital signal processors (DSP)—control embedded systems. Being committed to doing a certain task, which may need extremely powerful processors, is the crucial quality. For instance, even though they employ mainframe computers and specialized regional and national networks to connect airports and radar sites, air traffic control systems might be considered embedded.(Each radar most likely has a single or more embedded systems.)Design engineers may increase the embedded system's performance to maximum and reliability while reducing the product's size and cost because it is dedicated to certain functions. Embedded systems are sometimes mass-produced in order to take advantage of economics of scale. Physically embedded systems include everything from small, mobile gadgets like MP3 players and digital watches to big, fixed installations like traffic lights, factory controls, and nuclear power plant control systems.

A single microcontroller chip has a minimal level of complexity, while several units, peripherals, and networks housed inside a sizable chassis or enclosure have a very high level of complexity. Since most systems can be extended or programmed, the term "embedded system" is not one that can be defined precisely. For instance, mobile computers and embedded systems share some components, such as operating systems and the microprocessors that drive them, but handheld computers enable the loading of multiple applications and the connection of peripherals. Additionally, systems that do not disclose programmability as a core feature typically need to accommodate software updates. Large application systems will typically have subcomponents even if the system as a whole is "designed to perform one or a few dedicated functions," making it reasonable to label the system "embedded" on a continuum from "general purpose" to "embedded." [.1]

Communication deals with RF section and automation part deals with microcontroller ,dc geared motors, fire sensor ,line driver and battery. the project is explained as follows ,first signal from switches is transmitted to encoder (which converts digital signal to analog signal) and from there is transmitted to transmitter ,from transmitter it is transmitted to receiver of RF section and signal is decoded and decoded signal is given to controller(since controller accepts only digital signal).this signal is used for motion of robot. the fire sensor is used to sense the fire and provide signal to controller and controller controls the fire by using rolling water pump .a line driver is also present to reduce back currents and to produce voltage requieed for free running of dc geared motors.[2]

THEORITICAL SURVEY

The concept of electronic energy meters has been introduced in the power sector to effectively record the units consumed for billing purposes and also monitor several other factors to reduce power theft and minimize losses that occurs due to conventional electromechanical energy meters. This is a multipurpose project that integrates all the functions including a prepaid billing arrangement and automatic message sending system to the utility company.

EXISTING METHOD

Every home has an energy meter installed as part of the current system, which tracks energy usage. Then, a person employed by MSEB visits each and every home to gather the information, which he then provides to MSEB. The bill is then computed by MSEB .According to that, users receive their data bills by postal mail. After a considerable amount of time if a user doesn't pay the bill, MSEB sends staff to shut off that specific power supply .The MSEB sends a guy to connect the electricity supply when that person pays the invoice .Going to rural locations is difficult using this technology, for instance. Depending on who sends the MSEB, they may or may not be locals. It takes a long time to travel to isolated areas. That Not all users may take reading from a human.

When taking the reading, there is a chance for error. He then returns to submit data to MSEB. MSEB can then compute the outcome. Time is spent in this process in large amounts. This system requires a lot of working persons . MSEB must pay these workers extra for their job. A person goes to physically cut supply if MSEB must stop providing power. A person must return to the location where the power supply was initially unplugged in order to reconnect it. After the bill has been paid, MSEB sends a guy to connect the power supply. This method has various significant flaws, such as difficulty travelling to remote locations. [3]

Now a days we are using the postpaid energy meters that means the total number of units that we has been used ,for that we need to pay the bill .their was a line man for checking the meters and he can give the bill for the totalnumber of units that we has been using .For this they has been using the small bill reader to give full details that we has been used .that last date and the date of fine has been charged all this information will be indicated on the bill. [4]

Time is spent in this process in large amounts. This system requires a lot of workers or people. MSEB must pay these workers extra for their job. A person goes to physically cut supply if MSEB must stop providing power. A person must return to the location where the power supply was initially unplugged in order to reconnect it.

BLOCK DIAGRAM

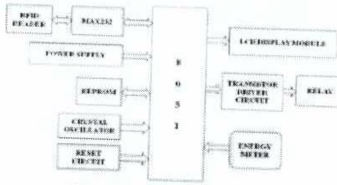
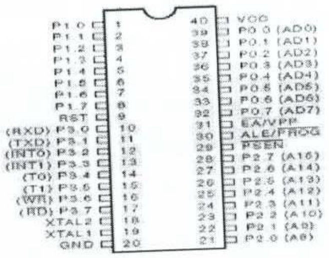


FIG.3.1 BLOCK DIAGRAM OF PREPAID ENERGY METER

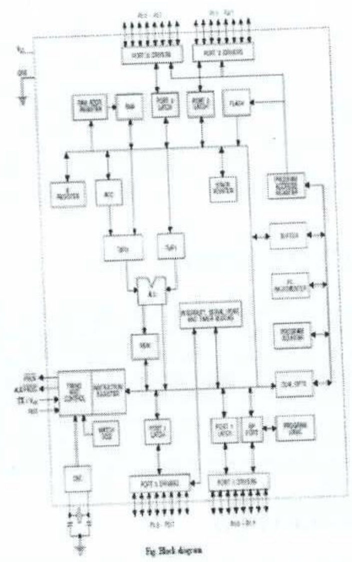
FEATURES OF AT89S52

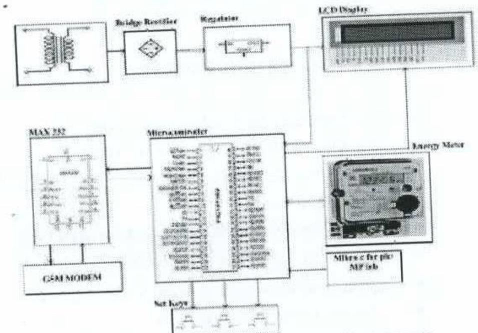
- The capacity of RAM is 256 bytes.
- The Operating Range is 4.0V to 5.5V
- 0 Hz to 33 MHz's is the Fully Static Operation range
- It has Three-level Program Memory Lock.
- It has 256* 8-bit Internal RAM.
- Programmable I/O Lines are 32
- 16-bit Timer/Counters -3
- Interrupt Sources-8

A high-performance, low-voltage CMOS 8-bit microprocessor with 8K bytes of Flash programmable memory is called the AT89s52. The product is made with high density nonvolatile memory technology from Atmel and works with the common MCS-51 instruction set. The on-chip flash enables in-system or external non-volatile memory programmer reprogramming of the programme memory. The Atmel AT89s52 is a potent microcomputer that offers a highly flexible and affordable solution to many embedded control applications by fusing a flexible 8-bit CPU with Flash on a monolithic chip. The AT89s52 also features two software selectable power saving modes and static logic allowing operation down to zero frequency.

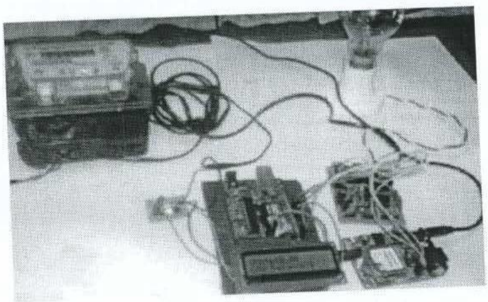


Block diagram of micro controller [4]





The further information and detail about this, the interfacing of prepaid energy meter with GSM modem system is available at website [illegible]



Working

A prepaid energy meter, also known as a prepaid electricity meter or a smart energy meter, is a device used to measure and control the consumption of electrical energy in a prepaid manner. It functions differently from traditional postpaid meters, where consumers are billed after using the electricity. Prepaid meters provide consumers with the ability to manage and control their energy usage in real-time, allowing for better budgeting and control over electricity expenses.

The detailed information about how this prepaid energy meter will work is explained below.

Meter Installation:

A prepaid energy meter is installed at the consumer's premises by the utility company. It consists of two main components: the metering unit and the customer interface unit (CIU). The metering unit measures the energy consumption, while the CIU serves as the interface for the consumer to interact with the meter. Token Generation: The utility company generates a unique token for each consumer, which contains information such as the amount of electricity units purchased and the duration of validity. This token is typically in the form of a numeric code or a smart card.

Token Input:

The token is entered into the CIU by the consumer. A smart card can be inserted into the CIU to accomplish this, or the code can be manually entered using a keypad.

The prepaid meter validates the token to make sure it is accurate and authentic. Decrypting the token and checking its integrity and authenticity using established methods and encryption keys are all part of the validation procedure.[4]

Credit Allocation:
Once the token is validated, the prepaid meter allocates the purchased electricity units as a credit to the consumer's account. The meter deducts the consumed energy units from this credit as the consumer uses electricity.

Energy Consumption Monitoring:
The metering unit continuously monitors the energy consumption in real-time. It measures the amount of electricity consumed by the consumer and maintains a running balance of the remaining credit.

Low Credit Warning:
When the remaining credit reaches a predefined threshold, the prepaid meter provides a warning to the consumer. This warning can be in the form of an audible alarm, a visual indication, or a message displayed on the CIU, alerting the consumer to recharge or top-up their account to avoid a power outage.

Power Disconnection:
If the consumer's credit balance is exhausted or remains low for an extended period, the prepaid meter initiates a power disconnection. It interrupts the electricity supply to the consumer's premises until a recharge or top-up is made.

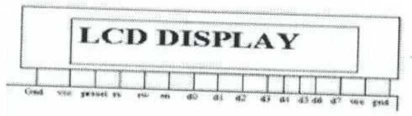
Recharge Options:
To recharge their account, consumers can purchase electricity units from authorized vendors or make online payments through designated platforms provided by the utility company. The consumer receives a new token containing the purchased electricity units, which can be inputted into the CIU to update the credit balance.[5]

Consumption History:
The prepaid energy meter keeps a record of the consumer's electricity consumption history. This data can be authorized and accessed by the utility company for billing purposes or by the consumer to track their energy usage patterns.

Prepaid energy meters has many advantages and benefits like better control over electricity usage, reduced chances of bill disputes, improved energy efficiency, and awake their customers about their energy consumption habits. They promote energy conservation and support it's customer or consumers to manage their energy expenses effectively.

Additionally, the Micro controller supplies the control signals to the LCD. Additionally, pins 3.5, 3.6, and 3.7 are used for this. By utilizing the port's bits in the programme the LCD is provided with the required control signals. If there is a need, the residual can be applied to another end. The programme completes the work for which it was created and controls the relevant ports. LCD interface functionality is provided by the software and related hardware..[6]

LCD interface.



Timer 2 Operating Modes

RCLK +TCLK	CP/$\overline{RL2}$	TR2	MODE
0	0	1	16-bit Auto-reload
0	1	1	16-bit Capture
1	X	1	Baud Rate Generator
X	X	0	(Off)

RESULT

The final output shows the consumption of electric power according to balance of recharge amount on the LCD display .Meter is recharged by GSM technology by sending the message to the module with some amount.

This system gives the buzzer alarm when recharge amount was depleted to RS03 units and it givesthe message to the user mobile as "Your available units are only 3".Further using of power leads to disconnection of power to the load when available units are zero and it gave the alarm and send a message "your available units are 00" and it displays please recharge on LCD display.

It is proven that this system can determine the existence of any warm person inside the coverage area and carry out subsequent actions based on a number of trials carried out This system can determine the existence of any warm body inside the coverage area and carry out subsequent actions, according to a number of studies carried out under diverse circumstances.

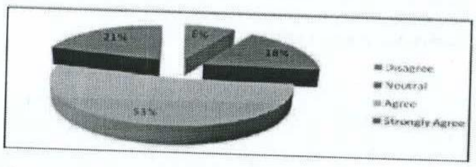


Fig pie chart of prepaid energy meter [6]

Future scope :

The future scope of this project is smart meters which uses internet and two way communication to transfer information and data between utility control and its consumers .so it reduces man power and saves electricity and reduces electricity bill

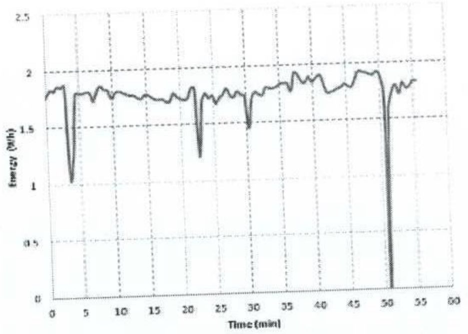


Fig: graph of energy verses time of internet based energy meter [7]

CONCLUSION

Based on several tests conducted, it has been demonstrated that this system can identify any warm individual inside the coverage area and take appropriate action .Due to the design of smart energy meters that make use of GSM technology, users may need to pay for electricity before using it. In this way, customers hold credit, use the electricity, and then use the leftover credit. Electricity is provided by a TRIAC section if the credit limit is reached. A plan is also made to notify the user via a GSM communication module when their credit balance falls below a specific threshold. This approach has been put forth as an original response to the issue of utility system affordability. The readings can be continuously recorded because the system is being developed using a microcontroller. This lessens the need for human labour while also improving the effectiveness of how electricity usage bills are calculated. Smart energy meters will offer a way to raise awareness about needless energy waste and will likely lead to less of it. By making connection easy to establish and preventing power theft, this module will lessen the strain of energy provision

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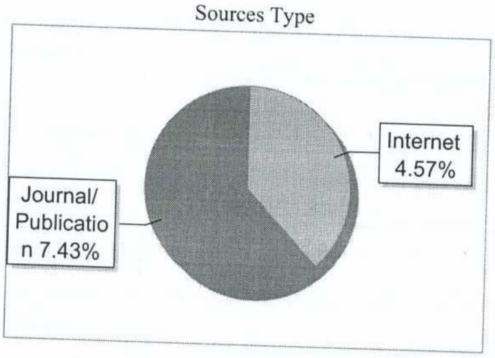

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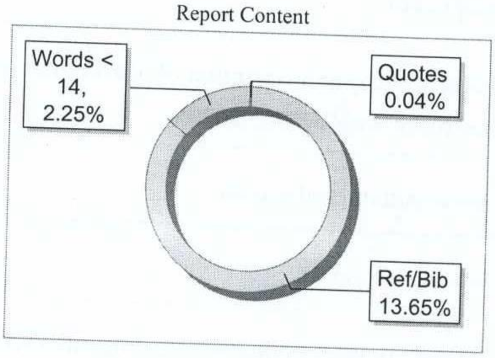

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MICROCONTROLLER-BASED SPEECH REALIZATION USING ALTERED CAT SWARM OPTIMIZATION

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Abstract:

Brain wellness can be monitored with the help of electroencephalography (EEG). Despite this, it is not often used outside of tertiary care because of a lack of facilities, competence, and patient preparation. Generally, the EEG data from humans are typically weak; it is, therefore, necessary to limit signal distortion, waveform offset, and other issues. EEG utilizes the electrical activity in brain cells. The EEG equipment records the electrical potential produced by activated cells. The cells transmit electric pulses to one another. Feelings facilitate interaction beyond words. Humans communicate best by speaking. Speech recognition in many countries is indeed a hot topic. In this paper, we propose a unique altered cat swarm optimization (ACSO) approach for the realization of optimal speech. Here, we design a hardware implementation of speech realization using the Microcontroller (MC) unit. Initially, temperature sensor and humidity sensor (EEG's primary sensor) signals are collected and these signals have some noisy/disturbance signals themselves also. To address this, a filter approach is utilized through the least mean square algorithm (LMSA). Arduino module is used only to transmit and receive such filtered signals. Then voice progression is taken by MC to generate the voice output. To optimize the voice output, the proposed ACSO technique is applied and then a voice module is employed to record the optimized signals. Moreover, an Internet of Things (IoT)/Wifi component is presented for storage purposes. The effectiveness of our work is assessed and compared throughout diverse contexts with the same loudspeaker to achieve the most effective voice messages. The findings are depicted by employing the MATLAB tool.

Keywords: EEG, Arduino, Microcontroller (MC), Internet of Things (IoT), Least Mean Square Algorithm (LMSA), Altered Cat Swarm Optimization (ACSO), MATLAB tool

I. Introduction

The feeling was a continuous as well as a crucial aspect throughout human life, despite the difficulties of exactly identifying such. A person's moods have a big impact on how they communicate, as well as how they behave & work [1]. Emotions are indeed important in day-to-day interaction. A term such as 'Okay' might be used to express happiness, sadness, or sarcasm.

Throughout several cases, the content of a conversation was deduced from the tone of the sound or nonverbal communication. Boredom, for example, is primarily indicated through body movements. Presently, computers and some other digital equipment are used for a substantial portion of the interaction [2]. That interaction differs significantly from how humans engage Non-verbal messages, account for the majority of human interactions, as well as the social factor of such an interaction is crucial.

Whenever individuals communicate through systems, they usually combine that social element as well. If a system detects the person's mood, such a relationship with or via a computer can be enhanced [3]. After computers were capable of recognizing feelings, several studies were conducted. Computers, for instance, were trained to identify emotions via voice, facial gestures, or a combination of the two. Emotion can be measured via brain function, which is a comparatively new technology. The EEG is indeed a simple and inexpensive way to evaluate brain function. EEG signals were discovered to contain emotional indicators.

Such signals have the benefit above other approaches in that they will be difficult to fool via voluntary movements and seem to be ready at all times without requiring the user to take some extra action [4]. The downside of using those signal would be that the person must carry a measuring system, which would be time-consuming. There are numerous options for this method. To begin with, the basic process of instantly identifying someone's feelings may aid clinicians & psychiatrists in conducting work. Additional uses include human-machine interaction, computer-mediated human communication, and supporting people with disabilities with emotional conversation.

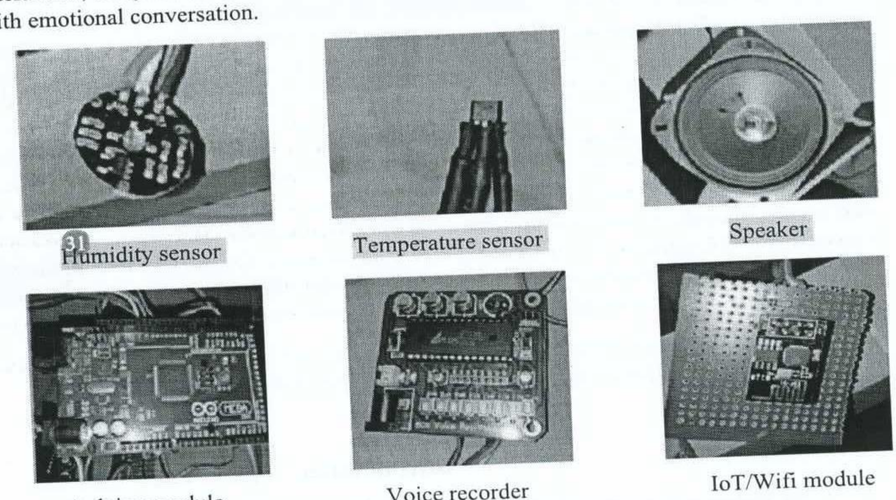

Figure 1: Components used in our research

Our main objective is to develop a hardware system that can identify emotions through EEG signals in real-world situations. In this paper, we propose the ACSO approach for effective speech realization. Figure1 depicts the utilized components of our research. Here, we design a hardware system of speech recognition by employing the MC unit. Initially, we receive signals from temperature and humidity sensors that are noisy/disturbed. To solve this, LMSA is utilized. The Arduino component only helps to exchange filtered data. MC then provides the speech output. The ACSO approach is applied to maximize the speech output, and then a voice recorder to capture the enhanced signals. In addition, IoT/Wifi is employed for storage. The additional part of this research is organized as topic II-literature survey and problem statement, topic III-proposed work, topic IV-performance analysis, and topic V-conclusion.

II. Related works

In this paper, we describe the previous research regarding speech realization as follows. Authors propose the Efficient Speech Recognition Engine (ESE) in [6], which is based on a compacted sparse Long short-term memory (LSTM) approach. ESE was improved throughout the algorithm-software-hardware divide: firstly, they introduced a technique for compressing throughout several cases, where the content of a conversation was deduced from the tone of the sound or nonverbal communication. Boredom, for example, is primarily indicated through body movements. Presently, computers and some other digital equipment are used for a substantial portion of the interaction [2]. That interaction differs significantly from how humans engage. Non-verbal messages account for the majority of human interactions as well as the social factor of such an interaction is crucial.

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LSTM model, without affecting the predictive performance, considerably reducing the memory capacity, can parallelize the complicated LSTM processes by mapping it to FPGA. The research [7] delves into the use of binary Deep Neural networks (DNN) in quick voice recognition inference. Firstly, effective binary matrix multiplication with hardware compatibility was designed based on bit activities like population number. A technique for computer-aided sleeping assessment was given in [8]. Tunable-Q wavelet transform (TQWT) has been used to deconstruct sleep EEG signal sections, as well as various statistical events which were retrieved and used as characteristics. During categorization, the Bootstrap aggregation has been used. The research shows that statistical moment-based characteristics may effectively represent the variations between distinct sleeping conditions. In [9], Automatic Speech Recognition (ASR) quality is attributed to noise, vibration, and Air conditioning (HVAC) sound levels. This lays the foundation for NVH & ASR designers to connect design attributes & effectiveness while selecting equipment that meets both organizations' goals. In [10], two-word length reduction techniques enabling estimated FFT construction are given and utilized on various FFT designs. The word lengths for such constructions were adjusted depending upon error & resource analyses. Including an energy efficiency of around 2.460 pico Jouls/neuron as well as a power usage of around 141.0 Watts, this study [11] offers an ultra-low power speech recognition system. The Binary convolutional neural network (BCNN) was

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enhanced via frame-level reuse, storage, partition, and weight normalization. A specialized estimate adder reduces latency while maintaining accuracy. That chip's on-chip self-learning compensates for the network binarization's precision reduction & gradually improves the detection capability towards a given user. In [12], Computer-aided spoken English learning relies upon voice recognition & assessment technologies. The in-depth learning technique is used to improve voice recognition performance & effectiveness. While the multiparameter approach has been used to evaluate characteristics that influence spoken speaking ability, the findings are also more compelling. Furthurmore, an empirical study was undertaken to assess the effectiveness of the assessment method based on computer voice recognition for oral English.

This paper [13] proposes an approach for artificial neural network (ANN)-based voice recognition using spike-timing-dependent plasticity (STDP) & local weight sharing, that is both medically unrealistic as well as energy-intensive. It succeeds at two voice recognition challenges. The spiking neural networks (SNNs) can acquire feature sets quickly and effectively, making voice information low-dimensional, sparse, as well as linearly distinguishable. Unlike global weight sharing, suggested, that local weight sharing was better in understanding voice signal attributes. The Raspberry Pi has been used as a control unit throughout this study [14], and the microphone has been used to gather voice recordings. In this case, the Baidu cloud voice recognition technique was deployed. The voice data could be used for voice command, debate, interaction, and remote car control. This could regulate the car's orientation, speed, as well as driving duration. The remote-control component enables us to communicate with the vehicle via LAN. This vehicle seems to have a whistling sound, an emergency stops as well as other features.

Researchers [15] present the double exponential adaptive threshold (DEXAT) neuron architecture, which increases the quality of the Recurrent Spiking Neural Networks (RSNNs) by allowing for faster convergence, and improved accuracy, as well as a flexible long short-term memory. They illustrate the DEXAT neuron component employing oxide-based non-filamentary resistive semiconductor switches that offer a hardware-efficient way for creating DEXAT neurons via tightly correlated circuit-device connections. This study [16] examines the recent studies in the field of smart voice recognition as well as the level of growth of the relevant hardware platform embedded systems. The study develops the general outline of machine learning within the Markov system based upon Markov random theory & machine learning theory, then founded on this concept, researches real-time speech vocabulary matching detection approach. An approximate computing approach regarding LSTM processes employed in voice recognition networks was given in this report [17]. This approach analyses the relevance of every LSTM cell during the execution & disabling duplicate LSTM cells based on a predefined similarity score. They also introduce additional cells that enable delta-based approximate LSTM computations with decreased resolution, called pseudo-skip procedures.

In [18], the authors try to acknowledge 10 Chinese numerical digits (1-10) using face myoelectric patterns. They created a circuit acquisition device with upper computer programs that capture user MES information from 4 voice streams. The moving average active portion has been extracted using threshold-based separation that improves the classification performance. The research [19] offers a voice recognition framework employing 2 neural network models. The Continous Hopfield structure classifies words independently addressing the speakers, but the backpropagation system detects the sound of four people. The primary goal was to create two techniques that can be used in a voice-activated safety system; however, the findings aren't secure sufficiently. An effective, uncontrolled, and scalable system enabling automated pattern recognition during live EEG surveillance was presented [20]. The key contributions were considerable: (1) combined time-domain characteristics were utilized to portray EEG data, revealing magnitude variation, particularly nonstationary EEG data. (2) Using RPM, real-time change detection was provided that could be applied with no previous training or understanding. In [21], the offline stress level identification system detects stress using EEG data. The LabVIEW programming effectively analyses EEG data as well as extracts the centroid value of Alpha & Beta bands to measure the level of stress. A unique method for detecting seizure beginning instances using multivariate EEG data was reported in this work [22]. It was revealed that graph-based properties related to that should describe the synchronized patterns of the epileptic 475 brain regions.

According to this study [23], structural time series analysis helps to determine changes in the EEG dataset. To detect the changes, the approach employs a nonoverlapping sliding window approach that quantifies the temporal similarity of EEG data. The concept's key way is to develop a distance measure that works with statistics. Pathology monitoring & categorization using IoT-cloud technology were proposed [24]. They employed two pretrained CNN architectures on such a standard EEG dataset. For the classifier, they used a pure time-domain EEG dataset, proving that end-to-end learning performs on an EEG database. Research [25] offers a noninvasive tool for determining the depth of anesthesia (DoA) using time-frequency assessment of EEG data collected over 15 streams with 10/20 electrode placement. Six subject EEG signals are captured & divided into thirty-second epochs having 50 percent overlapping. Every channel's time-frequency map (TFM) has been determined using a smoothed pseudo-Wigner-Ville distribution (SPWVD).

Problem statement:

Brain wellness can be monitored with the help of electroencephalography (EEG). Despite this, it is not often used outside of tertiary care because of a lack of facilities, competence, and patient preparation. Generally, the EEG data from humans are typically weak; it is, therefore, necessary to limit signal distortion, waveform offset, and other issues. EEG utilizes the electrical activity in brain cells. The EEG equipment records the electrical potential produced by activated cells. The cells transmit electric pulses to one another. Feelings facilitate interaction beyond words. Humans communicate best by speaking. Speech recognition in many countries is indeed a hot topic. Moreover, several existing works have been done on voice recognition by numerous algorithms. However, there is an insufficient outcome in voice ability, waveform signals, energy consumption, noise immunity, response time, etc. Thus, we present the ACSO approach for proper or efficient outcomes in speech realization.

III. Methodology

In this paper, we propose the ACSO approach for better speech realization. That means, we design a hardware system of speech recognition by employing the MC-module. Initially, signals from temperature and humidity sensors are gathered, which are noisy. To solve this, a filtering method called LSMA approach is utilized. The Arduino component only helps to exchange filtered data. MC then provides the speech output. ACSO algorithm is employed to maximize the speech output, and then a voice recorder to capture the enhanced signals. Additionally, IoT/Wifi is presented for storage. Figure 2 depicts the process flow of our proposed research.

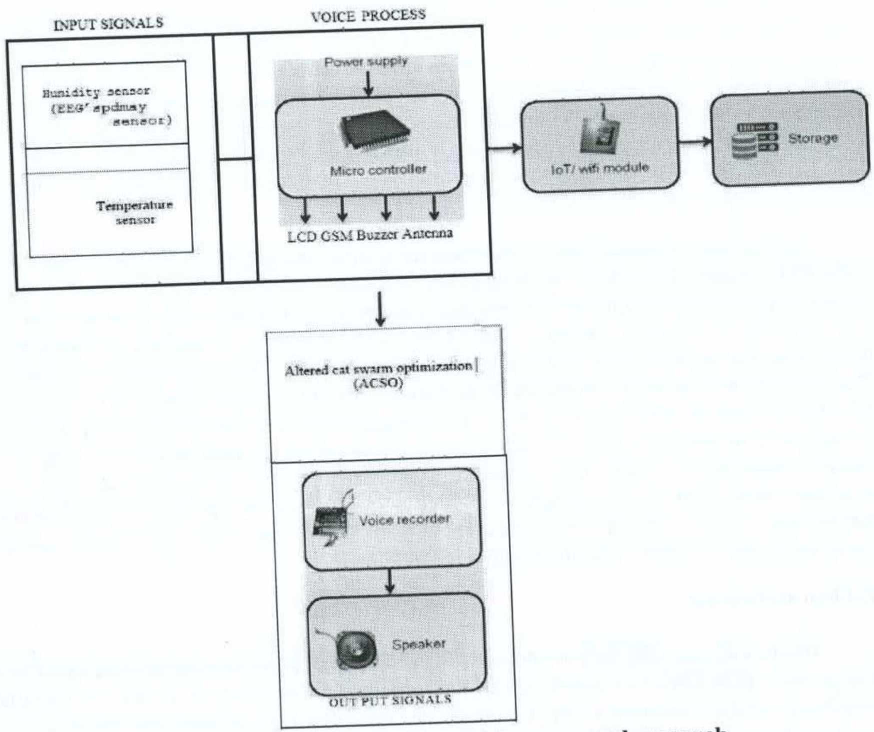

Figure 2: Procedure how of the proposed research

A. Adaptive FIR filter

Signal processing relies heavily on adaptive filters. Adaptive filters use an optimal technique to change their transfer function. Many adaptive filters have digitized filters owing to the complexity of the optimization techniques. Adaptive filters handle digital signals and adjust their output associated with the input signal. There are two types of adaptive filters: linear & non-linear adaptive filters. The non-linear adaptive filters contain higher difficult mathematics, although the linear adaptive filter was used in reality. The Adaptive FIR Filter's function block diagram is shown in Figure 3.

In this design, the input signal, variable filter response, and the desired signal are presented. The input signal is connected to the variable filter in this diagram, which produces an output signal. The error signal may reduce the error 14 by splitting the actual performance from the desired signal via altering the filter parameter.

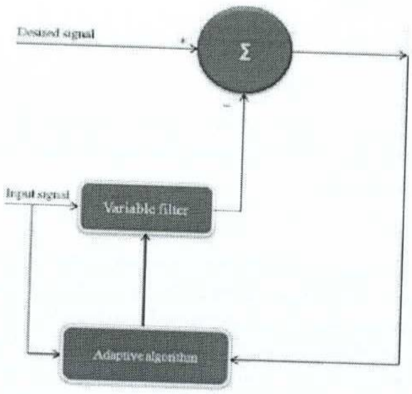

Figure 3: Adaptive FIR filter design

B. Filtration using least mean square algorithm (LSMA)

In this phase, we utilize the LSMA approach to amplify or standardize the gathered signals. From the captured signals, the distorted signal was created by adding the noise signal to the source EEG waveform. The LSMA approach is among the most widely employed adaptive filtering algorithms. Figure 4 depicts the diagrammatic representation of the LSMA method.

LSMA modifies the adaptive filter tap by such a greater percentage in the instant assessment of the gradient of an error function. It does not involve the computation of correlation function and/or matrix inversions, making it simpler and more efficient than traditional studies. The iterative approach contained inside to perform successive modifications throughout the direction of negativity of the gradient vector was described in the following calculations, resulting in the reduction of the mean square error.

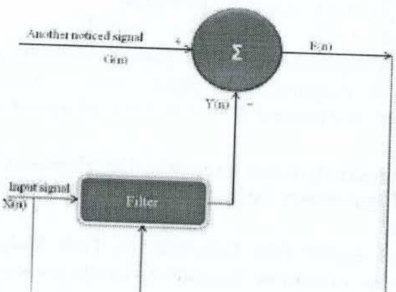

Figure 4: Schematic representation of LSMA

$$Y(n) = F(n) \,.\, U(n) \tag{1}$$

$$E(n) = G(n) - Y(n) \tag{2}$$

$$F(n+1) = F(n) + \mu \,.\, U(n) \,.\, E(n) \tag{3}$$

Here, X(n) = source signal, Y(n) = filter's output, G(n) = another noticed signal, and E(n) = error signal. Finally, we accomplish the signal from this stage with the noise-reduced outcome from the raw dataset.

C. Arduino platform

Individuals who are involved in semiconductors were becoming well associated with the Arduino platform. However, unlike the preceding programmable circuit boards, the Arduino does not always require an additional hardware device to load unique code onto the platform; instead, individuals could indeed easily use a USB connection to update, as well as the Arduino platform utilizes a simpler model of C++, enabling to understand to study & providing us with a much more accessible environment which bypasses the microcontroller's operations. The Arduino platform is divided into 2 sections:

➢ **Hardware section:** The Arduino board equipment is made up of numerous parts that work together to produce its operation.

i. USB-connection: That's the initial component of the Arduino and it is used to transfer a code towards the MC and seems to have a controlled the 5V supply that also powers the Arduino board.

ii. Additional power sources: It is being used to charge the board as well as offer a controlled voltage from 9V to 12V, which is useful if the USB connection is insufficient for whichever you have coded to perform.

iii. RESET switch: Whenever we hit the reset switch, the Arduino gets restored. This would be useful if you have loaded other instructions and just need the Arduino to execute them.

iv. Microcontroller (MC): It seems to be a system that collects and sends data or commands to the corresponding component.

v. Analog-port: A0 to A5 input terminals.

vi. Digital-port: 2 to 13 input/output terminals.

vii. In-Circuit Programmer (ICP): Another option for uploading our code would be to use the "TX" output & "RX" input.

viii. Power-port: We possess 3.30V & 5.0 V of power terminals.

➢ **Software section:** Arduino Integrated Development Environment (IDE) is indeed a collection of guidelines, which guide the equipment on what to do and how to accomplish it.

i. Command region: File, Edit, Sketch, Tool, Help, and symbols such as Confirm Symbols for validation, Transfer Symbols for loading the code, New, Open, Save, and Serial Monitors for transmitting and receiving data between both the Arduino as well as the IDE are all under that region.

ii. Content region: Here is where users create the coding, which would be written in a simpler form of the C++ language to help writing a code, usually known as a sketch, easier.

iii. Measure Window Region: Throughout the black region, we will get messages from IDE, primarily about code variations.

D. Altered cat swarm optimization (ACSO)

Cats and a cat behavior model were employed throughout the ACSO approach to reduce optimization issues, i.e., cats have been used to represent a set of solutions. Under ACSO, a choice must be made about how many cats will indeed be exploited, and afterward, the cats would be used to tackle issues within ACSO. Each cat seems to have its location, which would be made up of N dimensions, velocities for every dimension, a fitness value representing the cat adaption towards the fitness process, as well as a flag indicating if the cat was searching or tracing. Since ACSO preserves the proper result until it reaches the limit of iterations, the perfect outcome would have been the preferred place in one of the cats. Opposition-Based Learning (OBL) has been added in ACSO to increase searching capability.

➢ **Searching process**

The submodel has been used to represent a cat that would be resting, glancing around, as well as searching for a new place to move to. We describe 4 main parameters in the searching procedure: the searching memory pool (SMP), the searching range of the specified dimension (SRD), the counts per dimension to change (CDC), self-position consideration (SPC), as well as the mixing proportion (MP).

SMP: SMP has been used to determine the size of every cat's searching memory that represents the spots the cat is looking at. As per the criteria described later, the cat will select a spot from the memory pool.

SRD: The mutative proportion for such a specified dimension is declared by SRD. When a dimension has been specified to modify in the searching process, the variation among both the recent value as well as the previous value would not exceed the limit provided by SRD.

CDC: The CDC reveals how many dimensions would be modified. Every one of these elements plays a vital part throughout the search process.

SPC: SPC seems to be a Boolean factor that determines if such a cat's current location will be among the candidates for movement. The score of SMP would never be changed by whether the result of SPC was right or wrong.

MP: A concept termed mixing proportion (MP) is indeed a proportion of the population assigned a quite minimal number to ensure that such cats spend lots of time resting & observing, i.e., almost all of their time has been used in searching behavior.

The search mode's operation could be separated into five stages:

- Stage 1: Create k duplicates of the cat's current location, in which k equals SMP. When SPC indicates true, set k equal to (SMP-1), then the current location among the candidates.

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- Stage 2: As per the CDC, for every duplicate, substitute the previous values with fresh ones that are arbitrarily ±SRD percentage points of the current values.
- Stage 3: Determine each candidate spot's fitness values (F).
- Stage 4: When all F are not equal, use formula (4) to estimate the choice probability of every candidate point; alternatively, assign all choosen probabilities to 1.
- Stage 5: Choose a spot to relocate to at random from the candidate spots, and then update the cat's location.

$$Q_j = \frac{|F_j - F_c|}{F_{maximum} - F_{minimum}} \quad \text{where } 0 < j < k \tag{4}$$

$F_c = F_{maximum}$ if needed, fitness function's purpose seems to be to obtain the smallest response; else, $F_c = F_{maximum}$.

➢ **Tracing process:**

The training function is indeed a submodel that depicts a cat tracing certain objects. When a cat enters the tracing phase, it goes in all dimensions as per its velocity. The tracing mode's operation could be divided into 3 stages:

- Stage 1: Use formula (5) to modify the velocities in each dimension ($W_{l,e}$).
- Stage 2: Verify that the velocities are within the highest velocity range. If the modified velocity exceeds the range, assign this to the limit.
- Stage 3: Adjust the cat's location based on formula (14).

$$W_{l,e} = w_{l,e} + s_l \times d_l \times (y_{best,e} - y_{l,e}), e = 1,2,\ldots,N \tag{5}$$

Here, $y_{best,e}$ = location of the cat with the highest fitness value, $y_{l,e}$ = cat_l's location, d_l = constant, and s_l = random number between 0&1.

$$Y_{l,e} = y_{l,e} += w_{l,e} \tag{6}$$

The OBL strategy is introduced inside the ACSO technique to increase the diversity of the population. When Y is indeed the response to a specific issue, then perhaps the reverse of Y has been the other candidate solution, as per OBL. As a result, the probability of achieving an ideal solution gets improved.

Consider $y \in [b,c]$ and the reverse of Y will be

$$y' = b + c - y \tag{7}$$

The above-mentioned description could be expanded to greater dimensions.

Let $Q(y_1, y_2, \ldots\ldots y_m)$ = m- dimensional vector, where, $y_j \in [b_j, c_j]$ and j= 1,2,…..n.

The reverse vector of Q is described as

$$Q'(y'_1, y'_2, \ldots\ldots y'_m) \tag{8}$$

Where $y'_j = b_j + c_j - y_j$

➢ **Completion:**

If the completion criterion is fulfilled, then the procedure should be ended; else, the process should be repeated.

IV. Results & Performance Analysis

In this phase, the investigation of our research is provided to realize effective voice output. Table 1, depicts the hardware's specifications. Here, we utilize the sensed data (like temperature and humidity) as input signals. The signals are presented with certain noisy/unwanted signals. Then the filtering procedure was undertaken by the LSMA technique to generate the signals without noise signals. Arduino unit is employed to transmit and receive those filtered signals for voice processing. A microcontroller (MC) is utilized for voice processing and it contains several characteristics like LCD, GSM, and antenna.

The resultant signals from the MC unit have moderate efficiency. Therefore, the signals should be enhanced for proper voice recording by applying the proposed ACSO technique. That enhanced signals are documented in the voice recording element. Finally, such enhanced output signals are saved inside the databases (cloud) through employ the IoT (Wifi) module to evaluate the performance of the proposed technique. The performance metrics like realization efficiency, energy consumption, and mean square error (MSE) are examined to prove our research with the greater rate of voice realization.

Table 1: Hardware specifications

Components	**Design**
Temperature sensor	LM 35 4 volt-to-20volt range of operation 60 microampere maximum current
Humidity sensor	3.3 or 5-volt operating range 8-bit resolution
Arduino module	ATmega 2560 Rev3 Clock speed = 16 megahertz
Wifi component	ESP8266 ESP-01 1MB Flash Memory
Speaker	R= 4 ohm and P =5 W
Voice module	APR33A3 16-bit

The input signal and noise corrupted signal are depicted in Figures 5 and 6 correspondingly. In these graphs, the x-axis indicates time (sec) and the y-axis indicates frequency (Hz). The input signal should be presented with a noisy signal and it may degrade the voice signal. Thus, a filter must be utilized for this concern. The filtered signal is generated by using the LSMA approach as depicted in Figure 7.

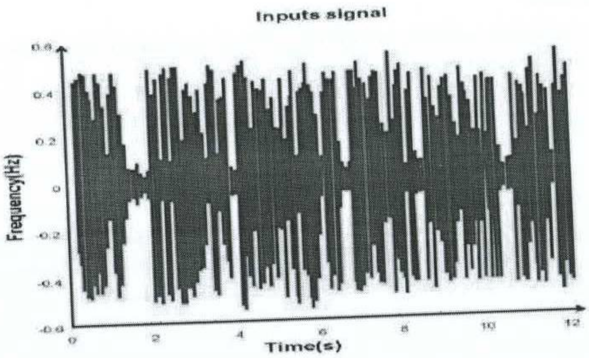

Figure 5: Input signal

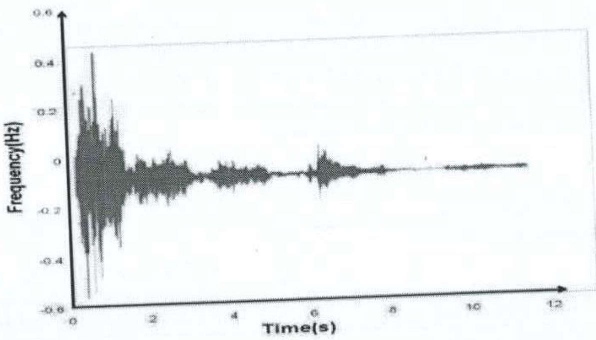

Figure 6: Noise corrupted signal

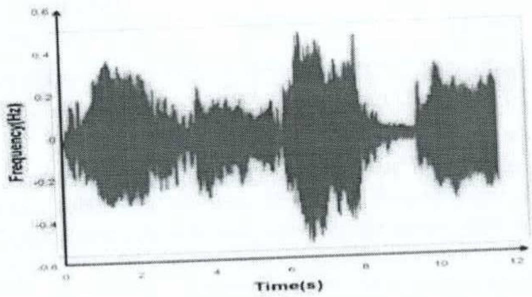

Figure 7: Filtered signal

Mean square error (MSE):

MSE is a technique for determining how closest estimations or predictions have become to actual numbers. Figure 8 depicts the MSE rate of the proposed technique for distinct conditions (happy, sadness, anger, and fear). By applying the equation (9), we effectively evaluate the error rate for our proposed work.

$$MSE = \frac{1}{m}\sum_{i=1}^{m}(s_i - q_i)^2 \tag{9}$$

Here, n= number of input signals, s = estimated score, and q = expected score.

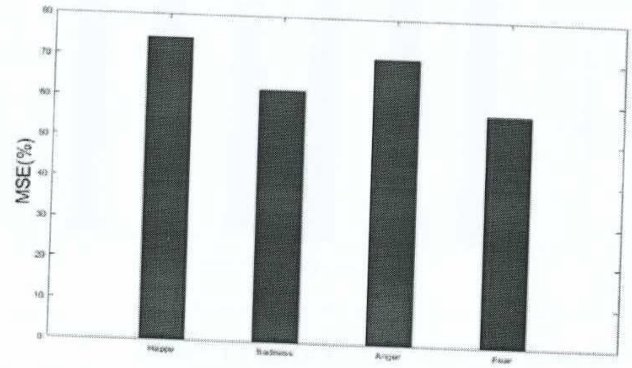

Figure 8: MSE for various conditions

Energy consumption:

The amount of electricity or power consumed is referred to as energy consumption. Figure 9 depicts the energy consumption rate of our proposed approach. In this graph, the x-axis represents the number of input signals and the y-axis represents the energy consumption. In this analysis, we used the lower percentage of energy consumption.

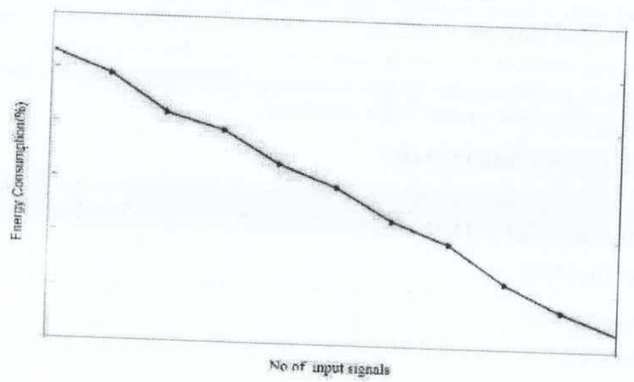

Figure 9: Energy consumption

Response time:

Responding to development in practical cases refers to the amount of time it takes from the input to which a task is assigned until it finishes its work. Figure 10 depicts the response time for our experiment regarding the number of input signals. From this graph, we accomplished our work in nearly 5.25 seconds.

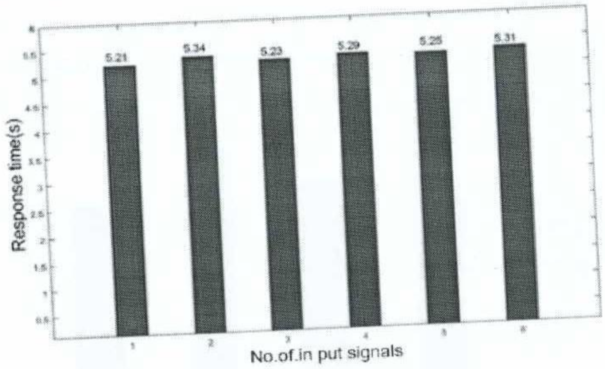

Figure 10: Response time

V. CONCLUSION

Moreover, numerous techniques were already developed for voice recognition. However, the results in voice quality, waveform signals, and energy usage are insufficient. This paper presents an ACSO approach to enhancing the quality of voice signals. That is, the microcontroller module was employed to design a hardware speech recognition system. Initially, we got noisy temperature and humidity sensor signals. To solve this, the LSMA filtering approach was utilized. The Arduino component just aids with data sharing. MC then provides the speech output. Then, the ACSO has been applied to boost the speech output and then recorded with a voice recorder. And the Wifi/IoT was used for storage purposes. Our presented technique provides outcomes in various conditions like happy, sadness, angry, and fear to prove our research with the greatest realization efficiency and reduced response time (nearly 5.25s). In the future, the CMOS technology is more efficient for certain features like minimal energy usage and greater noise immunity.

Declaration of Competing Interest

The authors declare that they have no known competing financial interests or personal relationships that could have appeared to influence the work reported in this paper.

Funding Availability

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Data Availability

No Data Availability

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