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Contents

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Hybrid Approach to Detect Prolonged Speech Segments

K B Drakshayini, Anusuya M A

A B S T R A C T

In the last 10 decades various methods have been introduced to detect prolonged speech segments automatically for stuttered speech signals. However less attention has been paid by researches in the detection of prolongation disorder at the parametric level. The aim of this study is to propose a hybrid approach to detect the prolonged speech segments by combining various spectral parameters with their recognition accuracies for the reconstructed speech signal. The paper presents prolonged segments detection by considering the parameters individually, combining various spectral parameters, validation of prolongation detection system, MFCC feature extraction process, basic model accuracies for the reconstructed signals. The proposed methods are simulated and experimented on UCLASS derived dataset. Obtained results are compared with the existing works of prolongation detection at parametric and word level. It is observed that hybrid parameters yield 92% of recognition rate for larger frame sizes of 200ms when modeled with SVM. The results are also tabulated and discussed for various metrics like sensitivity, specificity and accuracy metrics in detecting the prolonged segments. The study also focuses on the prolongation characteristics of vocalized and non-vocalized sounds at phoneme level. The detection accuracy of 71% is observed for Vocalized prolonged vowel phonemes over non-vocalized prolonged signal. Objectives: The objective of this work is to propose a hybrid algorithm to detect prolonged segments automatically for speech signal with prolongation disorder. The other objective is to evaluate the obtained spectral parameters performances by applying to various evaluation metrics and models to compute the recognition accuracy of a reconstructed signal. The objective is further extended to bring out the importance of variable frame size concept and to analyze the variations in vocalized and non-vocalized sounds. Methods: The methods adopted to detect prolonged speech segments are discussed at two levels namely at the preprocessing and modeling levels. The Preprocessing level is discussed by applying various parameters at an individual level, hybrid level by combing the Centroid, Entropy, Energy, ZCR parameters and MFCC feature extraction method. A new method has been applied using Specificity, Sensitivity and accuracy metrics to validate the prolongation detection model performance. In modeling level, the above parameters are discussed by applying evaluation metrics for the clustering and classification models like K-means, FCM and SVM. The performance of these methods is considered for evaluating and estimating the prolonged segment detection accuracy of the reconstructed speech signals of vocalized and non-vocalized sounds. All these methods are discussed in detail in the following sections. Findings: Hybridizing the spectral parameters to detect the prolonged speech segment automatically is a major finding of this work. It is also found that Specificity, sensitivity and accuracy metrics plays a major role in designing and validating the prolongation detection model. From the further experiments it is identified that the hybrid and verification metrics suits better for vocalized and non-vocalized sounds when larger frame lengths are considered. SVM has been found to perform better for all the above considerations. Novelty: As per Literature survey it is observed that individual and few parameters are applied to detect the prolongation. But works are not addressed on applying or combining more than two parameters to detect the prolonged speech segments. The novelty of this work lies in selecting and combining the spectral parameters at the preprocessing stage to detect the prolongation disorder. Spectral centroid and entropy are considered as appropriate parameters along with ZCR and Energy parameters. Hence hybridizing these parameters results in a novelty to propose an automatic prolongation detection system. Novelty is further brought by applying Specificity, sensitivity and accuracy metrics to build and evaluate the detection system for vocalized and non-vocalized prolonged sounds.

Keywords: Prolongation, Centroid, Entropy, Specificity, sensitivity, Autocorrelation, frame length,

Speech is the most common mode for communication to express ideas, feelings and thoughts [1]. There exists 2 to 5 % of population suffering from speech dysfluency. Stuttering is a break of normal speech such as repetition, prolongation and interjections of phonemes, sounds, phrases or word. It is required to identify the dysfluent part of speech automatically to enhance the process of speech signal analysis [2]. Few authors have worked by considering individual parameters for different datasets. In this Paper a hybrid method using Energy, Centroid, ZCR and Entropy is discussed to detect the prolonged dysfluent speech automatically. It also focuses on the performances of vocalized (for example: rrrrunning, aaaaple) and non-vocalized (for example: sssseven, ffffourteen) prolonged sound evaluated through various validation metrics. Recognition rate using various models are tabulated with their recognition performance. In the first level of preprocessing, prolonged segments are detected and removed. This signal is reconstructed to extract the MFCC coefficients and these features are modeled using various decision-making methods like KMeans, FCM and SVM are used to compute the recognition performance. These simulation results are analyzed and presented for various evaluation metrics at the parametric and the model levels. Section 1 and section 2 discusses about related literature and the methodology adopted.

Challenges and dataset used for simulation is presented in Section 3 and section 4. Section 5 discusses the results and observations. Conclusion and future enhancements are discussed in section 6.

1.1. Related literature

Table 1 depicts the related work available in the literature in brief using spectral parameters to detect the prolonged speech segments in a disfluent speech.

Table 1: Existing work on parameter-based prolongation detection process

Author	Database	Parameters	Decision making	Detection rate at parametric level	Recognition rate
Deshmukh [3]	UCLASS	Energy, pitch, duration, frequency formants	_	_	_
Katarzyna Barczewska[3]	6 recordings	First two formants (F1, F2), segment duration	_	_	68%
G. Manjula [4]	20 speakers (AIISH)	Epochs, Zero Frequency Filter	_	_	_
Sadeen Alharbi [4]	40 speakers (UCLASS)	ZCR and Short-term energy	_	70%	_
Waldemar Suszyński[5]	10 speakers	Frequency and duration	Fuzzy classifier	91%	_

II. METHODOLOGY

Figure 1 depicts the complete architecture of the proposed system. The architecture is considered at three phases. The First phase is aimed at preprocessing the signal to reduce the noise and to detect the prolonged segments automatically by using the proposed hybrid parameterization technique. In the second phase the reconstructed signal is further processed to extract first level 12 speech features using MFCC procedure. In the last phase these features are modeled using various models to compute speech recognition accuracies.

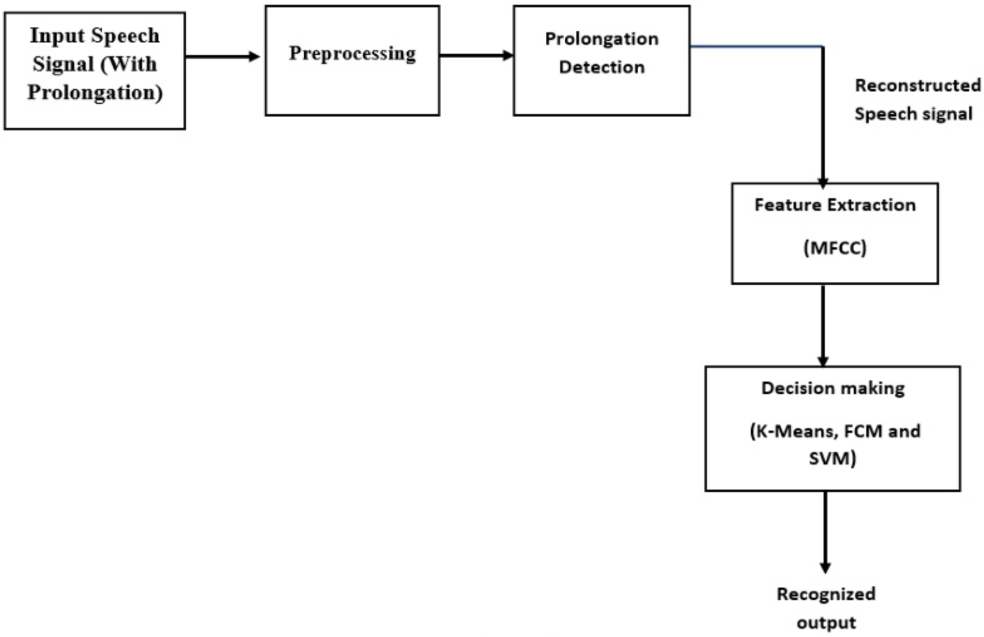

Figure 1: Stuttered Speech recognition Process

2.1. Phase I: Prolongation detection

Phase 1 consists of preprocessing prolongation detection and reconstruction of the signal.

2.1.1 Preprocessing:

The original sampling frequency of the input speech samples of UCLASS dataset is 44.1 kHz, speech samples are down sampled to 16 kHz for the simulation purpose. In the digitization process the speech signals are normalized to prevent the error estimation caused by change in speaker's volume.

2.1.2 Parameters considered for prolongation detection

To detect the prolonged speech segments the following parameters are adopted and evaluated individually. These Parameters are also combined to observe the performance of hybrid effect in the prolongation detection.

A. Short term Energy

Let , n = 1…, N the speech samples of the I th frame, of length N. Eqn. (1) used for computing short term energy of an individual frame. The stress on a particular phoneme has higher energy associated with it [6]. Hence short-term energy is one of the parameters to detect prolongation in stuttered speech signal.

$$E_i = \frac{1}{N}\sum_{n=1}^{N} |x_i(n)|^2 \text{----- Eqn. (1)}$$

B. Spectral centroid

The spectral centroid represents centre of gravity of speech spectrum and it is highly variated for speech segments [6]. It also indicates the centre of mass spectrum and depicts the brightness of sound. It is calculated using Eqn. (2)

$$C_i = \frac{\sum_{k=1}^{N}(k+1)x_i(k)}{\sum_{k=1}^{N}x_i(k)}.x_i(k) \text{-----Eqn. (2)}$$

k = 1..., N, are the Discrete Fourier Transform (DFT) coefficients of the i^{th} frame, where N is the frame length.

C. Zero Crossing rate (ZCR)

ZCR is a measure of smoothness of a speech signal and represents frequency at which the power is focused in the signal [7]. Zero crossing rate is computed using Eqn. (3)

$$Z_n = \sum_{\infty}^{-\infty} |\,\mathrm{sgn}[x(m)] - \mathrm{sgn}[x(m-1)]\,|\; w(n-m) \text{ -----Eqn (3)}$$

Where

$$\mathrm{sgn}[x(n)] = \begin{cases} 1, & x(n) \geq 0 \\ -1, & x(n) < 0 \end{cases} \text{-----Eqn (3a)}$$

and

$$w(n) = \begin{cases} \frac{1}{2N} & for\ \ 0 \leq n \leq N-1 \\ 0 & for, otherwise \end{cases} \text{-----Eqn (3b)}$$

N is the window duration.

D. Spectral Entropy

Spectral Entropy calculates the regularity of power spectrum of speech signal and has peak capturing property [8]. Distinct peaks and the position of the peaks in the spectra are dependent on the phoneme under consideration. It is computed using Eqn (4).

$$Entropy = \frac{-\sum_{k=b_1}^{b_2} s_k \log(s_k)}{\log(b_2 - b_1)} \text{-----Eqn (4)}$$

where

- sk is the spectral value at frame k.
- b1 and b2 are the band edges, over which spectral entropy is calculated.

Similarity measure: Autocorrelation

Autocorrelation is a measure of similarity between the successive frames. Highly correlated frames are considered as prolonged frames. To differentiate between prolonged and un-prolonged speech segments autocorrelation similarity measure is applied over adjacent frames of the signal [9]. Autocorrelation function is computed by Eqn (5).

The Autocorrelation between the samples of X= {x_1, x_2, …,x_n} and Y= {y_1, y_2, ….,y_n} is given by the correlation factor and it is given by

$$r_{XY} = \frac{\sum_{i=1}^{n} (x_i - X')(y_i - Y')}{\sqrt{\sum_{i=1}^{n}(x_i - X')^2}\sqrt{\sum_{i=1}^{n}(y_i - Y')^2}} \text{-----Eqn (5)}$$

where X' and Y' are the mean values of X and Y respectively.

All the above parameters are computed for the total length of the signal on frame-by-frame basis. From our observation it is observed that these parameters play a major role in differentiating between prolonged and normal speech frames. The simulations results are discussed at two levels by considering parameters individually and by combining the parameters. The recognition accuracies for these two levels are calculated and discussed in section 5.

Observation from literature:

In the literature researchers have applied only short-term energy and ZCR parameter among the above discussed parameters to detect the prolonged segments, whereas spectral centroid and Entropy are not. But ZCR parameter is applied for dysarthric dysfluent speech but not for stuttered. In this work ZCR, Energy entropy and centroid parameters are applied together to detect the prolonged frame. To detect the prolonged segments the proposed algorithm is as follows:

Algorithm to detect and remove prolonged frames

Step 1: Read the prolonged stuttered speech signal
Step 2: Divide the signal into frames of 0.025ms and 200 ms duration.
Step 3: Compute Short term Energy, ZCR, Spectral Entropy and centroid parameters for each frame to the complete signal length.
Step 4: Autocorrelation function is applied to compute the similarity between the adjacent frames.
Step:5 Average autocorrelation values for each parameter are computed.
Step 6: Every parameter is identified and fixed with the threshold values.
Step7: If the autocorrelation value between the adjacent frames is greater than the frames are identified as prolonged speech frame and it is removed.
Step8: Retained frames in steps 7 are identified as un prolonged frames and used to reconstruct the speech signal.
Step9: steps1 to 7 are repeated by the individual spectral parameter threshold value on each frame at a time to decide the segment is prolonged or not.

Figure 2 depicts the hybrid method by combining all the parameters to detect the prolongation detection.

Average Autocorrelation of these parameters are represented by the following terms. Short-term energy (E-ACF), Autocorrelation of ZCR (Z-ACF), Auto correlation of Entropy (En-ACF), Autocorrelation of centroid (C-ACF). These are used to compute the similarity measures in terms of spectral parameters. The average autocorrelation values of each parameter are considered as Threshold. Threshold values of the above-mentioned parameters are computed using autocorrelation function and these terms are defined as Autocorrelation threshold for Energy (TE), Autocorrelation threshold for ZCR (TZ), Autocorrelation threshold for Entropy (Ten), Autocorrelation threshold for Centroid (TC). Using the above-mentioned threshold values the decisions are made to retain or to remove the frames. Retained frames are combined to reconstruct the fluent speech signal.

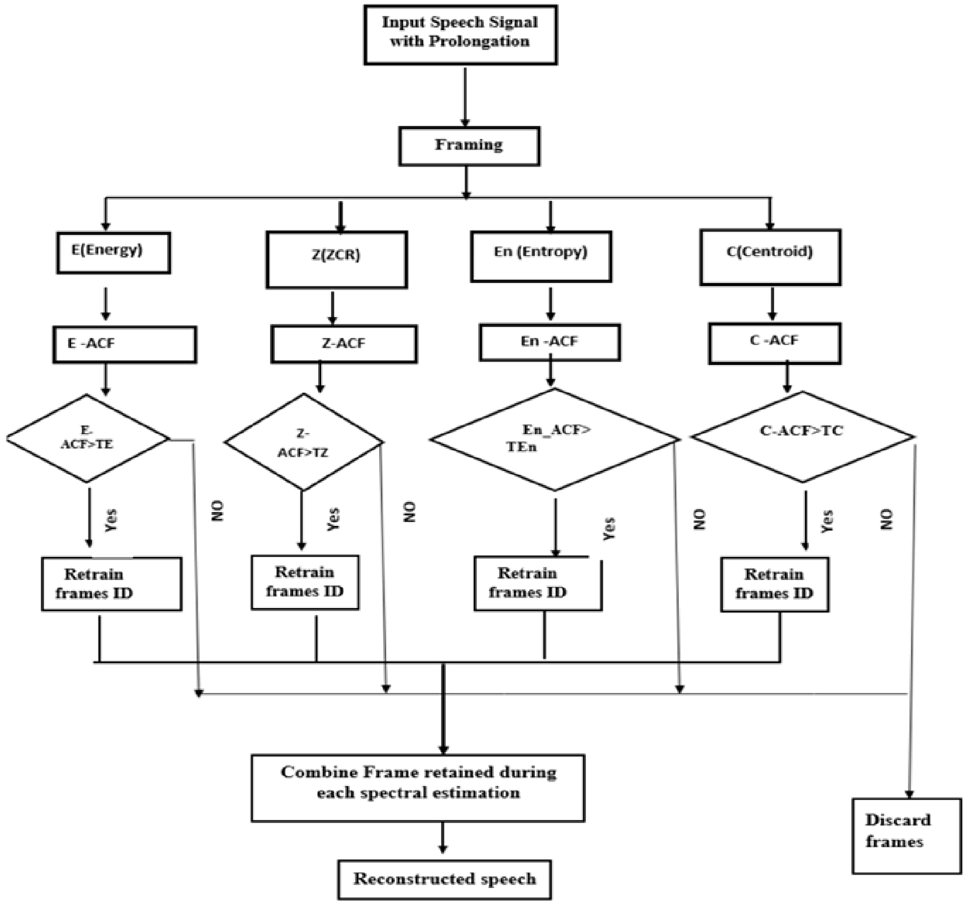

Figure 2: Prolongation Detection Process

2.2 Phase II: Feature Extraction using MFCC

The features of the reconstructed signal are computed by applying the MFCC feature extraction procedure. The first 12 coefficients of MFCC are computed for each frame. Below algorithm list the steps used to extract the speech parameters: Algorithm for feature extraction is as follows

Step 1. Each of the speech sample is down sampled to 16 KHz to increase the quality of the signal and passed through high pass filter to remove the noise.
Step 2. The pre-emphasized signal is framed in to 200 ms and 0.025ms duration with an overlap of 10ms
Step 3. Hamming window is applied to smoothen the signal and Discrete cosine Transform is applied.
Step 4. Compute the Log filtered energies and first 12 coefficients of the signal using IDFT for all the frames.
Step 5. Average featured frames are computed.

Observations:

The advantage of considering larger frame size helps in better identification of the prolonged frames that majority checks all the parameters threshold values. It also reduces the number of frames to be processed with steady or low SNR regions. Larger frame size also helps in loosing less prolonged data.

2.3 Phase III: Modeling Phase/Decision:

The obtained features from the above process are modeled using K-Means, FCM and SVM classifiers to calculate the recognition accuracy of the reconstructed signal. The following section discusses the procedure to calculate the recognition rate for various models. This work considers two clustering methods and one classifier to compute the recognition performance of the reconstructed speech signal. The basic settings of the models for the simulation to compute the recognition accuracy is as discussed below.

2.3.1 K-means: K-means algorithm is used to generate a vector quantization codebook for data compression. It partitions 'N' observations into K clusters, in which each observation belongs to the cluster with the nearest mean. Each of clusters is defined by its central vector or centroid. Using Euclidian distance function, K-Means algorithm clusters the data in to K groups that assigns objects to their closest cluster [2].

Algorithm:

Step1.Set Observations (N=50) and clusters (K=5).
Step2. Randomly select 5 distinct centroids for each cluster.
Step3.Using Euclidian distance measure to find the distance between each observation point and the centroid.
Step4.Assigns each observation point to the nearest cluster
Step5.Calculate the mean of each cluster as new centroid
Step6.Repeat step 3 to step 5 with the new center of cluster until there is no change in the centroid.

2.3.2 FCM (Fuzzy C Means)

This algorithm works by assigning a membership value to each data point corresponding to each cluster center on the basis of distance between the cluster center and the data point [11]. The performance of the algorithm depends on membership value(U), and the fuzzy parameter(m), termination criteria, norm of the matrix,

Algorithm

1. Randomly initialize the clusters (K=5)
2. Create the distance matrix from a data point to each of the cluster center using Euclidean distance
3. The Membership matrix is computed using fuzzification parameter m= (2.0)
4. Values of the membership matrix should be less than or equal to one otherwise return to step2
5. Termination parameter is set as 0.0001

2.3.3 SVM (Support Vector Machine)

SVM is a powerful machine learning tool which attempts to obtain a good separating hyper-plane between two classes in the higher dimensional space. It is a predominant technique to estimate the basic parameters of speech. Speech sample can be approximated as a linear combination of speech sample by minimizing the sum of squared difference between the actual speech samples and predicted values. A unique set of parameters or predictor coefficients can be determined. These coefficients form the basis for linear prediction of speech samples. The importance of this method lies in its stability to provide extremely accurate estimates in understanding nonlinear phase characteristics of stuttered speech[10].

Algorithm:

Step 1. Best hyperplane is decided by considering the distance between data and exiting hyperplane.
Step 2. To make decision on deciding optimal hyperplane, polynomial kernel function with degree 4 is to decide optimal hyper plane with the hold out factor 0.15.
Step3.To classify frames into different classes multiclass classification is broken down into multiple binary
classification. Classifier uses m (m=5) binary SVM classifiers.
Step 4: In each binary SVM classifier, member ship is predicted to make decision of belongingness.

III. CHALLENGES

Deciding frame length is a challenging task because it has to accommodate all parametric variations of phonemes and also it should not break the disorder part of the speech segment while framing. If the prolongation duration extends, it is difficult to analyze and append smaller frames in terms of prolonged and un-prolonged phonemes. It is also an equally challenging task to classify prolonged speech sound categories that are mixed with other types of disorders. Fixing the threshold values to each type of the sounds is tedious a task, that to when combined with nasal sounds in non-vocalized words.

IV. DATASET

The experiments are conducted on the UCLASS dataset. This repository consists monologs, readings, and conversational recordings [12] [13] [14]. For our simulation 80 samples at word level are derived from the sentences recorded in the repository. It includes 22words of female speakers and 58 words of male speakers with age ranging from 11 years to 20 years. The samples are chosen to cover speech samples of different age and gender. By perception, vocalized and nonvocalized prolonged words are identified and derived manually. A total of 80 speech recordings of prolongation with vocalized and nonvocalized sounds are collected and few examples are depicted below. Table 2 depicts example words considered for vocalized and non-vocalized prolonged sounds used in our simulations.

Table 2: Sample Data set considered for experiment

Actual word	Pronounced Pattern	Type of Prolonged sound
Ball	/b/aaaaa/ll	Vocalized
Came	/c/aaaaa/me	
climbed	/c/laaaaa/imbed	
Every	/eeee/very	
Moment	/mmm/oment	
Favourite	/fffff/avourite	Non-vocalized
Finding	/fffff/inding	
Fish	/ffffff/ish	
Step	/ssss/ta/eee/p	
Secondary	/ssss/econdary	

V. RESULTS AND DISCUSSIONS

This section discusses the simulation results for prolongation detection and recognition accuracies obtained for the prolonged stutter speech. The results are discussed at four levels as listed below:
i) Prolongation detection using individual parameters
ii)Prolongation detection by combining/hybridizing spectral parameters I
ii)Performance evaluation of prolongation detection using validation metrics at parametric level
iv)Recognition accuracy of the reconstructed speech using models.

5.1 Prolongation detection using individual parameters

This section discusses the results and observations individually by considering all the spectral parameters to detecting the prolonged speech segments.

Short term energy:

The Short-term energy and autocorrelation between the adjacent frames are computed using Eqn (1) and Eqn (5). To distinguish between prolonged and un-prolonged frames the average threshold value (TE) is set as 1. 43e06db.If autocorrelation of frame energy value exceeds 1.43e06db it is detected as prolonged else it is un-prolonged. In the Prolonged frames partly or completely the energy of two adjacent frame will be approximately same due to the prolongation of certain phoneme. The threshold value changes as the type of the phoneme and sound changes. For Example, for the speech signal pronounced as name where phoneme 'a' is prolonged 'n/aaaa/me'. The experiments are conducted for two frame sizes by varying the frame length. For 200ms and 0.025 ms, total 4 and 17 number of prolonged frames were identified. For example, considering 200 ms, obtained total 10 frames, out of which one set of adjacent frames (7th and 8th frames) are identified as prolonged frames having short term energies as 1.17e03db and 2.26e03db respectively. The difference between autocorrelation of these two is computed as 4.8e06. Since value is greater than the threshold value 1.43 e06it is considered as prolonged frame. Figure 3a shows short term energy of a prolonged speech signal pronounced as "n/aaaa/me ".

Zero crossing rate

The ZCR and autocorrelation between the adjacent frame are computed using Eqn (2) and Eqn (5). To distinguish between prolonged and un-prolonged frames average threshold value (TZ) is set as 1. 5e04.If autocorrelation of Zero crossing rate value exceeds 1.5e04it is detected as prolonged else it is un-prolonged. In the Prolonged frames partly or completely the ZCR of two adjacent frames will be approximately same due to the prolongation of certain phoneme. The threshold value changes as the type of the phoneme For Example, for the speech signal pronounced as name where phoneme 'a' is prolonged 'n/aaaa/me'. The experiments are conducted for two frame sizes by varying the frame length. For 200ms and 0.025 ms, total 4 and 17 number of prolonged frames were identified. For example, considering 200 ms, obtained total 10 frames, out of which one set of adjacent frames (7th and 8th frames) are identified as prolonged frames having 97and 271 number of zero crossings respectively. The difference between the autocorrelation of these two is computed as 6.4 e04. Since this value is greater than the threshold value 1.5e04 it is considered as prolonged frame. Figure 3b shows Zero crossing rate of a prolonged speech signal pronounced as "n/aaaa/me ".

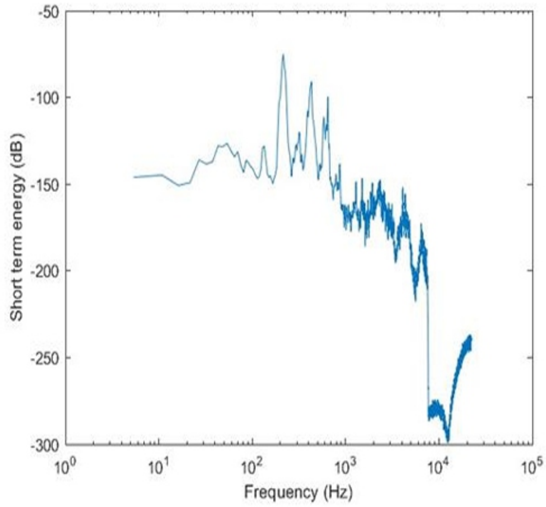

Figure 3a: Short term energy contour of a speech signal

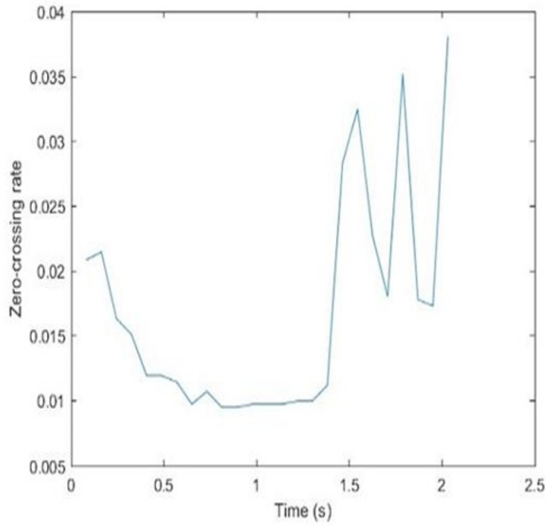

Figure 3b: Zero crossing rate of speech signal

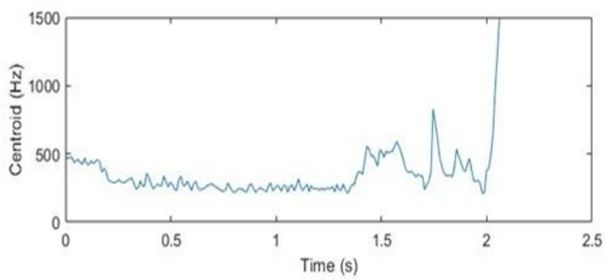

Figure 3c: Spectral Centroid of a Speech signal

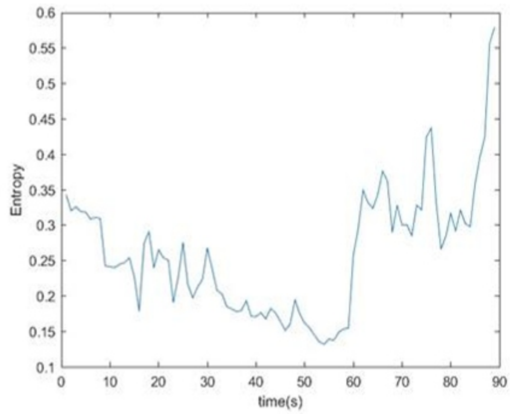

Figure 3d: Spectral Entropy of a speech signal

Spectral Centroid

The centroid and autocorrelation between the adjacent frame are computed using Eqn (3) and Eqn (5). To distinguish between prolonged and un-prolonged frames average threshold value (TC) is set as 0.0015 Hz. If autocorrelation of Spectral centroid value exceeds 0.0015 Hz it is detected as prolonged else it is un-prolonged. In the Prolonged frames partly or completely the Centroid of two adjacent frame will be approximately same due to the prolongation of certain phoneme. The threshold value changes as the type of the phoneme and sound changes.

For Example, for the speech signal pronounced as name where phoneme 'a' is prolonged 'n/aaaa/me'. The experiments are conducted for two frame sizes by varying the frame length. For 200ms and 0.025 ms, total 4 and 17 number of prolonged frames were identified. For example, considering 200 ms, obtained total 10 frames, out of which one set of adjacent frames (8th and 9th frames) are identified as prolonged frames having spectral centroid values 0.07 Hz and 0.05 Hz respectively. The difference between autocorrelation of these two is computed as 0.0059 Hz. Since this value is greater than the threshold value 0.0015Hz it is considered as prolonged frame. Figure 3c shows Spectral Centroid of a prolonged speech signal pronounced as "n/aaaa/me ".

Spectral entropy

The Spectral entropy (Shannon) and autocorrelation between the adjacent frame are computed using Eqn (4) and Eqn (5). To distinguish between prolonged and un-prolonged frames the average threshold value (TEn) is set as 1.1e05Hz. If autocorrelation of Spectral entropy value exceeds 1.1e05Hz it is detected as prolonged else it is un-prolonged. In the Prolonged frames partly or completely the entropy of two adjacent frame will be approximately same due to the prolongation of certain phoneme. The threshold value changes as the type of the phoneme and sound changes.
For Example, for the speech signal pronounced as name where phoneme ‘a’ is prolonged ‘n/aaaa/me’. The experiments are conducted for two frame sizes by varying the frame length. For 200ms and 0.025 ms, total 4 and 17 number of prolonged frames were identified. For example, considering 200 ms, obtained total 10 frames, out of which one set of adjacent frames (5th and 6th frames) are identified prolonged frames having spectral entropy values 0.07 Hz and 0.05 Hz respectively. The difference between autocorrelation of these two is computed as 2.4 e05 Hz. Since this value is greater than the threshold value 1.1 e05Hz it is considered as prolonged frame. Figure 3d shows Spectral Centroid of a prolonged speech signal pronounced as “n/aaaa/me “.

Observations:

Sometimes if the signal has been shortly pronounced with the minimum of one or two frames identifies results in negative Entropy values, which is not considered. The observed result for prolongation detection with respect to individual parameters is tabulated in Table 3.

5. 2. Prolongation detection using hybrid method

From the above discussions it is realized that all the parameters are required and plays a major role in detecting the prolongation at frame level in one or the other way. Hence work motivated to observe the combined effect of these parameters, for a frame length of 200ms. Since larger frame size can accommodate all the variations of the parameters the major discussions are done for the 200ms frame duration. To present the performances difference between smaller and larger frame results are tabulated in table 3 for both the size of frames for 200 ms and 0.025ms.

From the above experimental observation, it is clear that all these parameters play a major role in identifying strength of the signal hence work motivated to combine all these parameters and to propose a new hybrid prolongation detection algorithm. The above parameters are applied for variable frame length analysis to increase the prolongation detection rate. The results are tabulated prolongation detection rates for 200 ms and 0.025 ms frame duration in Table3 and analysis is shown in Figure 4.

Table 3: Prolongation detection process results (with frame duration 0.025msec and 200 msec for speech signal name.wav pronounced as ‘n/aaaa/me’)

Parameter	No of proper frames detected	No of prolonged frames detected	% of detection	No of frames detected as Proper frames	No of frames detected as prolonged	% of detection
Frame duration	0.025msec (Total 82 frames)			200ms (Total 10 Frames)		
Energy	65	17	20.73	6	4	40
ZCR	72	10	12.19	5	5	50
Entropy	56	26	31.70	5	5	50
Centroid	61	21	25.60	4	6	60
Proposed Hybrid method	40	42	51.21	3	7	70

From the above table it is clear that hybrid method yields better prolongation detection rate than individual parameters for larger frame size. This improvement is because, it considers the frames that are left out with individual parameter due to smaller frame size. The threshold values that lie on the frame boarder is not neglected due to the larger frame sizes during framing which can happen in smaller size. This fails to locate prolonged frames when individual parameters are used. But the above variations are overlooked in the hybrid approach.

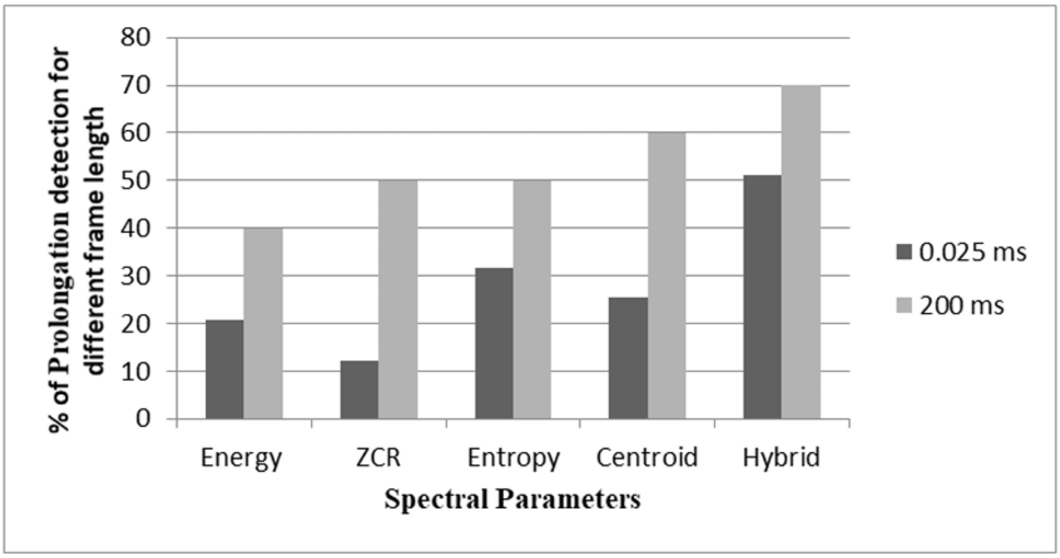

Figure 4: Analysis of proposed prolongation detection method with variable frame length for individual parameters and hybrid method

5.3. Performance Evaluation of Prolongation Detection Using Validation Metrics

Further to ensure these selected spectral features are of high prominence, the verification and validation of these spectral features performance is evaluated by considering Specificity, accuracy and sensitivity parameters as defined as follows: This section presents the measures to evaluate the appropriate selection of the spectral parameters in detecting the prolonged frames in terms of sensitivity, specificity, and accuracy measures. As per the Table 4 the hybrid method performs well. The performance metrics are computed using Equations 6 ,7 and 8. These metrics are defined in terms of TP (true positive), TN (True negative), FP (false positive) and FN (False Negative). The validation results are tabulated in table 4 and Analysis is shown in Figure 6.

Specificity: The specificity of a prolongation detection system is to determine the proper speech frames correctly. Mathematically, this can be stated as follows.

$$\text{Specificity} = \frac{TN}{TN+FP} \text{ ----Eqn (6)}$$

Sensitivity: The sensitivity of a prolongation detection system is to determine the prolonged speech frames correctly. Mathematically, this can be stated as follows.

$$\text{Sensitivity} = \frac{TP}{TP+FN} \text{ ----Eqn (7)}$$

Accuracy: The Accuracy of a prolongation detection system is to differentiate the prolonged speech frames and proper frames correctly. Mathematically, this can be stated as follows.

$$\text{Accuracy} = \frac{TN+TP}{TN+FP+TP+FN} \text{ ----Eqn (8)}$$

Where

- True positive (TP): Prolonged frames correctly identified as Prolonged frames
- False positive (FP): Proper frames incorrectly identified as Prolonged frames
- True negative (TN): proper frames correctly identified as Proper frames
- False negative (FN): Prolonged frames incorrectly identified as Proper frames

Table 4: Comparison of Performance evaluation metrics for different spectral parameters during prolongation detection (Frame duration: 200ms, 10 frames)

Parameter	TP	TN	FP	FN	Specificity	Sensitivity	Accuracy
Energy	4	6	3	4	66	50	58
ZCR	5	5	3	3	62	62	62
Entropy	5	5	3	3	62	62	62
Centroid	6	4	3	3	57	66	62
Proposed Hybrid method	7	3	1	3	75	70	71

Observations:

From the above table the prolongation detection model has accuracy measure as a major metric to differentiate between prolonged and un-prolonged frames. As an extension, interested observations were made for vocalized and non-vocalized sounds by taking accuracy metric to detect the prolonged speech segment. Sample vocalized and non-vocalized sounds analysis for prolongation detection and correction is as shown in Figure 5(a)-5(d) and Table 5.

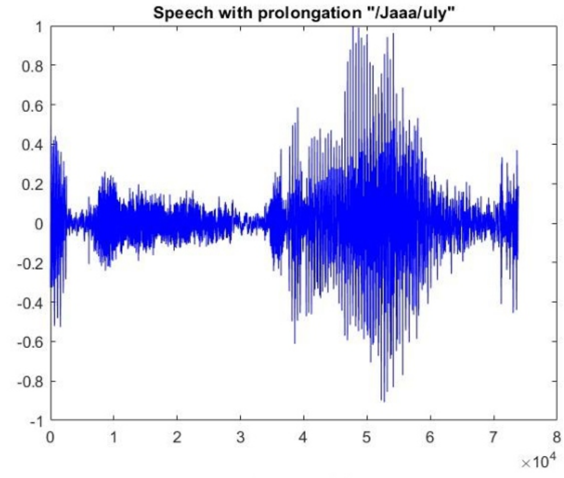

Figure5(a)

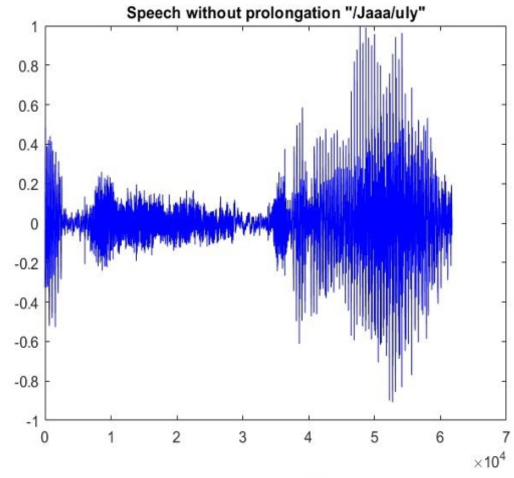

Figure 5(b)

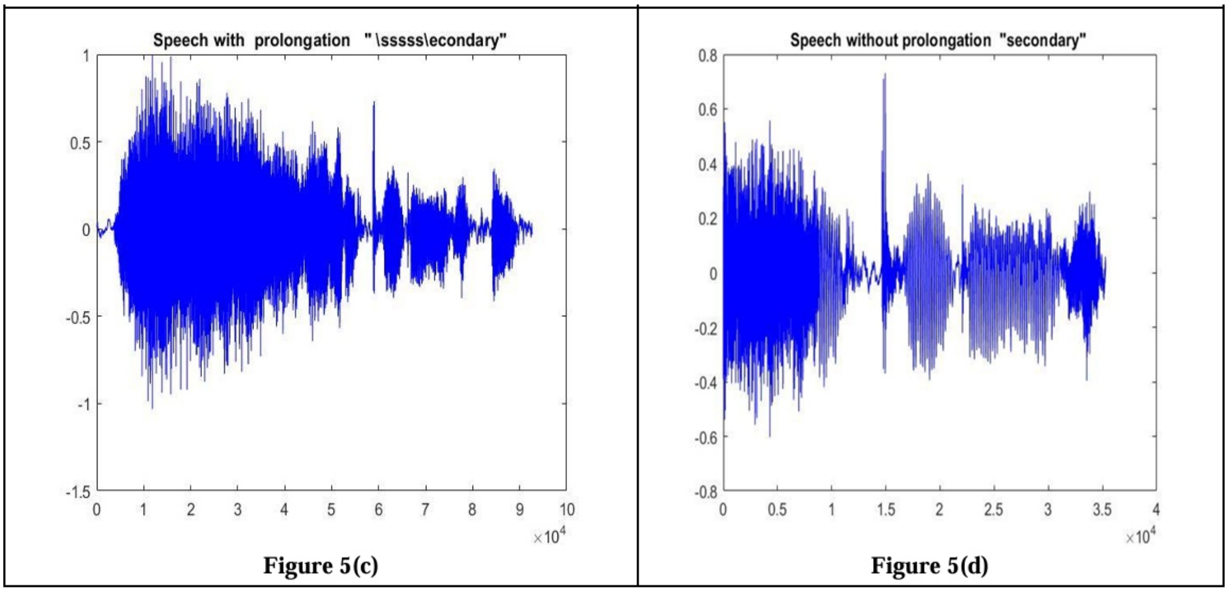

Figure 5: Vocalized and non-vocalized prolonged sounds before and after removal of prolongation.

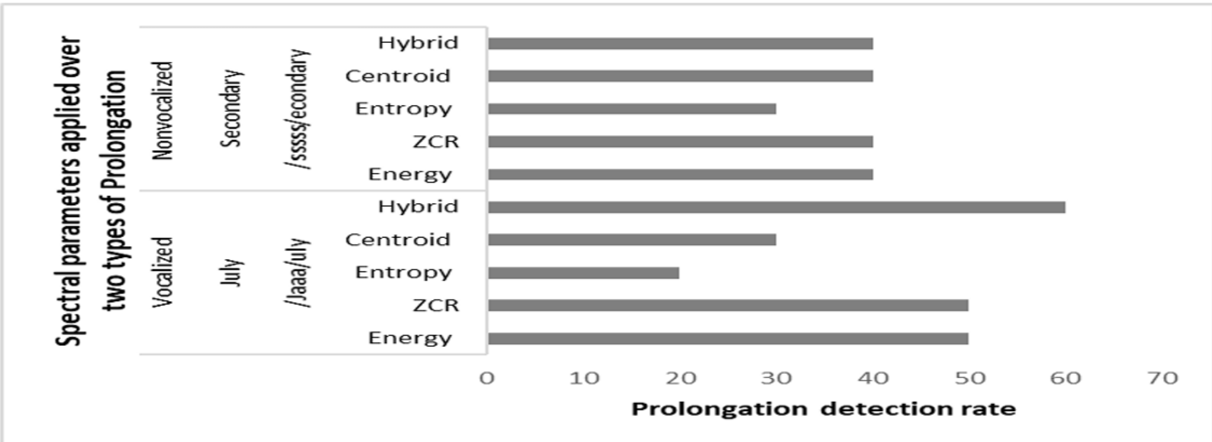

Figure 6: Analysis of proposed prolongation detection method for vocalized and nonvocalized prolongation (Frame duration 200 ms)

Observations:

In detecting vocalized prolonged sounds vowel sounds are clearly detected with major contributions of ZCR and Energy parameters. In prolongation of vocalized sound ‘Jaaa/uly ‘(Actual word: July) ‘a’ is detected as prolonged phoneme. During detection process prolonged speech sample is divided in to 10 frames with frame duration of 200 msec. Out of 10 frames,6 frames are identified as prolonged frames. non-vocalized prolonged sounds, intersection of voiceless consonants in between the phonemes could not find regularity in power spectrum due to the poor prolongation. In prolongation of non-vocalized sound ‘sssss/econdary ‘(Actual word: secondary) ‘s’ is detected as prolonged frame in non-vocalized prolongation audio sample (Ex: ‘sssss/econdary’) prolongation detection process prolonged speech sample is divided in to 10 frames with frame duration 200 msec. Out of 10 frames, 4 frames are detected as prolonged frames by combining results of all 4 spectral parameters.

5.4. Performance of Stuttered Speech Recognition System:

The signal is reconstructed by removing the prolonged frames through hybrid procedure. Features are extracted using MFCC procedure. First 12 coefficients are extracted and modeled by applying K-means, FCM and SVM classifiers. Table 5 depicts the recognition accuracies of all models and analysis is shown in Figure 7. Since the hybrid method was well suited to vocalized sounds, the simulations were performed for vocalized sounds. Among the above models SVM has obtained 92% of recognition accuracies for the reconstructed signal. From this it is clear that hybrid parameters contribute towards the detection, correction and recognition of vocalized sounds than non-vocalized sounds in a prolonged stuttered speech.

Table 5: Effect of prolongation detection method in stuttered speech recognition

Parameter used for Prolongation detection		Energy	ZCR	Entropy	Centroid	Proposed Hybrid
Feature Extraction		MFCC				
Decision making	K-Means	65%	68%	65%	70%	80%
	FCM	68%	70%	68%	72%	85%
	SVM	70%	72%	70%	75%	92%

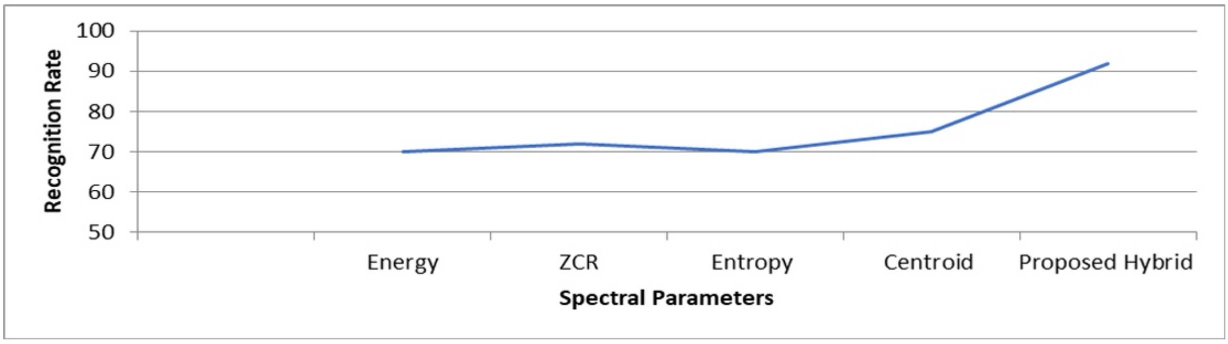

Figure 7: Analysis of effect of proposed prolongation detection method for stuttered speech recognition model

VI. CONCLUSIONS AND FUTURE ENHANCEMENT

This work proposes a hybrid approach to detect and recognize prolonged speech segments using spectral parameters. The prolongation detection is discussed for variable frame lengths of vocalized and non-vocalized prolonged sound. It is observed that 71% of prolongation detection and 92% of recognition accuracy is identified for the proposed hybrid approach over the method. It is also observed that the hybrid approach is less suitable for non-vocalized sounds. Hence the approach can be further enhanced to increase its efficiency for non-vocalized sounds by defining various threshold measures for phoneme sounds and by identifying few more parameters by trial-and-error methods. Further it can be tried by different entropy parameters over non-vocalized sounds. The work can be further extended to phrase level to detect the prolongations at word level by increasing and considering the data set for various age and gender parameter.

DECLARATION

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Ethical Approval and Consent to Participate	No, the article does not require ethical approval and consent to participate with evidence.
Availability of Data and Material/ Data Access Statement	Not relevant.
Authors Contributions	All authors have equal participation in this article

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Video Summarization: A Review on Local Binary Pattern and Classification Process

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A B S T R A C T

Video summarization system can yield good results if the high level features also called the semantic concepts in video frame are modeled accurately by considering the temporal aspects of the frames. The existing system is context aware surveillance video summarization which is a Domain dependent System. It works only on low level features and correlation between them is extracted and updated using dictionary algorithm in an online fashion. Thus dictionary size increases. In contrast to the existing method, the proposed system is a domain adaptive video summarization framework based on high level features in such a way that the summarized video can capture the key contents by assuring minimum number of frames. One of the high level features extracted is Local binary pattern (LBP).Key frames can be extracted after finding the Euclidean distance between the LBP descriptor in different methods. The key frames are classified using k-means clustering algorithm. The result is compared with several datasets thus showing the effectiveness of the proposed system. The entire work can be simulated using matlab.

Keywords — Euclidean distance; feature extraction; LBP; video summarization

I. INTRODUCTION

There is a huge growth in video data that calls for an urgent need to develop tools that summarize events occurring in these videos. Large parts of most videos are often redundant or not informative. So manually watching for hours only to figure out the informative events is very time consuming. Furthermore, it is difficult for people to focus on watching videos for hours and not miss important events in the video. So, it is very important to develop tools that automatically select the most informative parts of a video sequence. This summarization technique is called Video Summarization. Video summarization is divided into two:

a) Static Video Summarization It is also called static video storyboard, that contains a set of key frames which is extracted. from the original video

b) Dynamic Video Summarization It is also called dynamic video skimming that computes the similarity or relationship of each shot by collecting a set of shots.

In this paper, a simple approach for video summarization is proposed in a domain adaptive framework which is based on extraction of high level features from video frames and are classified using any learning techniques.
In addition, a new methodology for evaluating video summarization by comparing with VSUMM dataset and user summaries is considered for comparison. Thus evaluation of VSUMM is performed on different videos e.g.: cartoons, news, sports, tv-shows) etc.

II. LITERATURE SURVEY

Shu Zhang, Yingying Zhu, and Amit K. Roy-Chowdhury [1], proposed a method called Context-Aware Surveillance Video Summarization. There are two main algorithms used in this method. One is sparse group lasso optimization algorithm and other one is Online updates the dictionary of correlation algorithm. This method is mainly focused on summarizing surveillance videos. Features and correlation that exist among features of individual video frames are also considered.

Zhuang et al. (1998) [2] proposed a method using unsupervised clustering for key frame extraction. Here, the video is segmented into shots and then a color histogram is calculated for every frame. The clustering algorithm uses a threshold that controls the clustering density. Before a new frame is classified, the similarity between the node and the centroid of the cluster is computed first. If this value is less than threshold, then this node is not close enough to be added into the cluster. The key frame selection is employed only to the clusters which are considered as key clusters. In that case, a representative frame is extracted from this cluster as the key frame. The key frame is selected as the frame which is closest, to the key cluster centroid for each key cluster. This proposed technique is efficient and no comparative evaluation is performed for validating such assertions

Hanjalic and Zhang (1999) [3] proposed a method for producing a summary of an arbitrary video sequence which is based on cluster-validity analysis and is designed to work without any human supervision. This entire video material is first grouped into clusters. Then each frame is represented by color histograms in the YUV color space. Now, a partitional clustering is applied n times to all frames .Then the prespecified number of clusters starts at one and is increased by one each time the clustering is applied. Thus, the system automatically calculates the optimal combination of clusters by applying the cluster-validity analysis. After this optimal number of clusters is found, each cluster is represented by one characteristic frame, which becomes a new key frame. Hanjalic and Zhang (1999) concentrated on the evaluation of the proposed procedure for cluster- validity analysis, rather than on evaluating the produced summaries.

Gong and Liu (2000) [4] proposed a technique for video summarization based on Singular Value Decomposition (SVD).

Firstly, a set of frames in the input video is selected. Then, color histograms in the RGB color space are used to represent video frames. Each frame is divided into 3 * 3 blocks, and a 3D-histogram is created for each of the blocks to incorporate spatial information. Then, these nine histograms are concatenated together to form a feature vector. A feature-frame matrix A (usually sparse) is created for the video sequence using this feature vector extracted from the frames, Then, SVD is performed on A to obtain the matrix V, in which each column vector represents one frame in the feature space. Then, the cluster closest to the origin of the feature space is found, and then the content value of this cluster is computed. This value is used as the threshold for clustering the remaining frames. Now from each cluster, the system selects the frame that is closest to the cluster center as key frame.

III. PROPOSED METHOD

Fig 1: shows the steps of our method that produces a domain adaptive static video summary. Firstly, the original video is split into frames and then high level features are extracted, i.e. here Local binary pattern (LBP) features are calculated and are classified using an unsupervised method called k-means clustering. In most of the existing methods, domain dependent system is used for summarization (for e.g. only: surveillance videos, sports videos etc.).

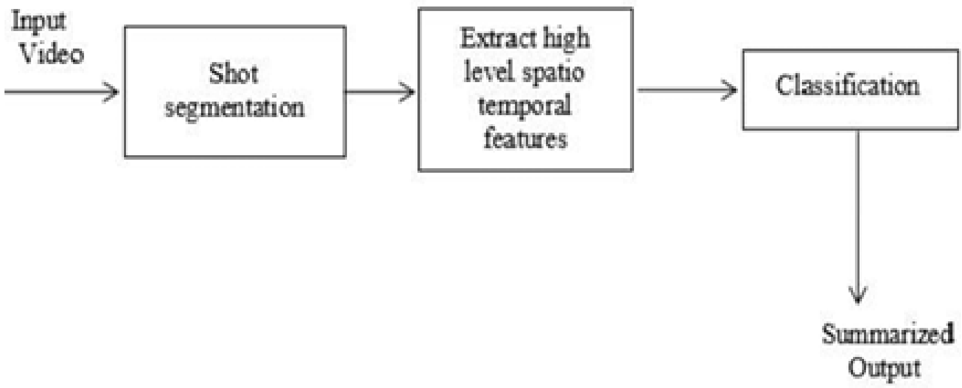


Fig 1: Block diagram for proposed method

VSUMM approach doesn't take all frames. So frame rate is calculated and corresponding frames are taken for feature extraction. Then these frames are classified to extract key-frame. The meaningless frames are removed from video sample. Then, by using k-means clustering the frames are grouped. By calculating Euclidean distance between frames of each cluster and within each cluster, one frame per cluster is selected which the selected frame is called Key frame. Now similar key frames are eliminated to refine static video summary. Finally, remaining key frames are arranged in temporal order.

Steps involved are:

3.1. Shot Segmentation

Here, the video stream is split into shots or frames of images. Frame rate is calculated for every video. This method is also called pre-sampling approach where sampling rate is fixed on one frame per second. For e.g.: normal frame rates are 24fps, 30fps, 60fps etc.

3.2. Feature Extraction

In the existing methods, low level spatio temporal features are extracted to detect motion regions and to detect multiple events. For e.g.: features like spatio-temporal interest point (STIP) detector, histogram of oriented gradients (HOG) and histogram of optical flow (HOF) features are extracted to
detect motion regions. But this will not give an accurate result after extracting features. So a high level spatio temporal feature is extracted here.

Local binary pattern (LBP), which is a visual used for classification in computer vision. It's mainly used for texture classification. When LBP is combined with (HOG) descriptor, it improves the detection performance on datasets. LBP feature vector is created as:

• Firstly, divide the examined window into cells (e.g. 16x16 pixels for each cell).
• Now for each pixel in a cell, compare the pixel to each of its 8 neighbours. Follow the pixels along a circle, i.e. clockwise or counter-clockwise.
• When the center pixel's value is greater than the neighbor's value, set binary array to "0" else "1". This gives an 8-digit binary number (which is usually converted to decimal).

• Optionally normalize the histogram.

• Now concatenate (normalized) histograms of all cells which gives a feature vector for the entire window.

The feature vector can now be processed using the Support vector machine or some other machine-learning algorithm to classify images. Such classifiers can be used for face recognition or texture analysis. Here it is processed using means clustering algorithm.

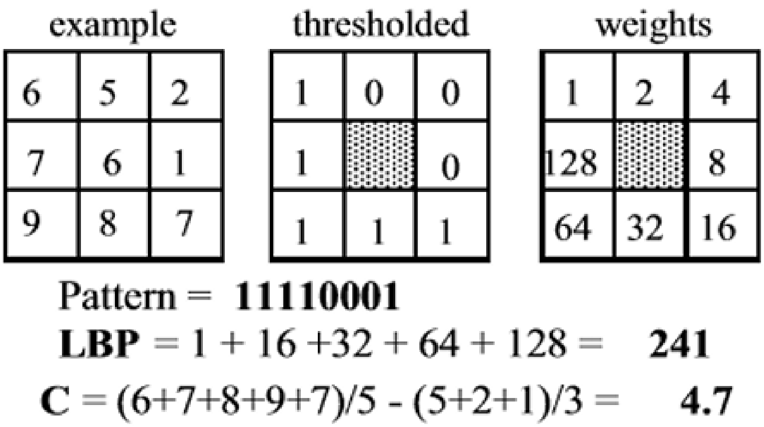


Fig 2: Calculation of LBP values

3.3. Clustering Technique

Clustering is a method of grouping similar frames within a cluster or in between clusters. In the existing methods, a High density peak search (HDPS) clustering algorithm and a Video representation based high density peak search (VRHDPS) clustering algorithm was used for integrating some important properties of video.

In this paper, the most efficient clustering method when compared to existing method is K-means clustering algorithm. This is one of the simplest methods of unsupervised learning algorithm. In this work, k-means clustering is applied to frames extracted using LBP feature descriptor. Now, Euclidean distances between LBP features are calculated and then classified using k-means clustering algorithm. For finding the centroid of each cluster, Euclidean distance between the clusters and within the clusters is calculated. Thus for each key cluster, the closest frame to the centroid cluster which is measured by Euclidean distance is selected as key frame. The value of k is user defined. When the value of k increases, redundancy will occur. If it decreases, there will be loss of key frames. So threshold value (i.e. k value) is set according to this. But different videos have different k value. This is the main drawback of k-means clustering.

IV. EXPERIMENTAL RESULT

The experiments were performed on VSUMM dataset which includes VSUMM summaries and user summaries of videos to improve the effectiveness of video summarization. VSUMM dataset contains several videos with different events for e.g.: cartoons, news, sports, TV-shows, home videos etc.

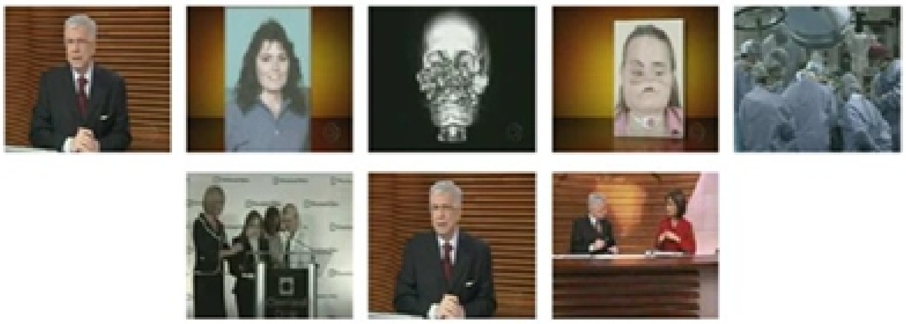
Surveillance videos are also considered for summarization since they have lot of redundancy. A comparison with VSUMM dataset and user summary is performed on extracted frames after clustering. If these frames exist in the summary, then that frame is a key frame else redundant frame. Thus redundant or useless frames are removed. Experimental set up is done for more than 50 videos each with a duration varying from 1 to 10 min.

Fig 4: shows VSUMM summary and one of the user summaries of cartoon video. The extracted frames are compared. But result shows key frames are missing and no: of redundant frames increases when k value increases. Say for k=10, there are missing frames as well as useless frames. So accuracy is affected. Thus future scope depends on getting accurate result with minimum redundancy and maximum key frames.

Thus an alternate method to summarize the key frames is using LBP extraction and setting a threshold value i.e. mean or standard deviation instead of using any learning algorithms like k-means clustering for classification process. This results in better accuracy and performance when compared to k-means clustering process.

Fig 3: Result analysis of k means clustering for k=10

(a) VSUMM



(b) User #5

Fig 4: Dataset of VSUMM summary and one user summary of a cartoon video

V. CONCLUSION

Video Summarization has attracted a fast growing attention from researchers and thus several algorithms and techniques have been proposed. In this work, a review on domain adaptive framework using LBP and k-means clustering is carried out. An alternate method using threshold value calculation is also done for comparison and to produce better result. VSUMM dataset is used to produce static video summaries. The evaluation process includes comparison between VSUMM dataset, user summary and extracted key frames of video. Thus, this technique produces video summaries of high visual quality and also can be used for summarization of different types of compressed videos. Video summarization produces more informative summary if semantic features are combined with visual Future work is based on overcoming the drawback of k-means clustering technique and this can be extended any other semantic feature as well as different clustering algorithm.

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Analysis of OFDM based V-band RoF Millimeter Wave System with Wireless AWGN Channel

Chandu C.B., Suparna Sreedhar A., Nandan S.

ABSTRACT

As technology advances, the bandwidth requirement is increasing day by day. These growing needs push the RF carrier frequencies towards the millimeter wavebands. A radio over fiber link schemes with Quadrature Amplitude Modulation (QAM) with 5 Gbps data rate in OFDM format is presented in this paper. The 60 GHz millimeter wave is generated by implementing Frequency Quadrupling techniques. Optical generation method is adopted in this paper. The central office to base station link as well as the base station to mobile station links is analysed with the help of simulation software.

Index Terms: OFDM, Radio over Fiber, Full duplex, RF down conversion

I. INTRODUCTION

The last few decades can be considered as an era of development in wireless communication techniques. The requirement of broadband access is one of the ever increasing needs in the present scenario. The researchers face a major challenge to satisfy this increased bandwidth need of the users [1]. One of the efficient solutions to this problem is to move towards the millimeter wave bands. The designation given by International Telecommunication Union to this band is Extremely High Frequency Band (EHF). The electromagnetic spectrum varies from 30-300 GHz in this band.

High atmospheric attenuation is one of the main problems occur to the signals in the EHF frequency band. This attenuation result in a reduced range of about 1 Km for the terrestrial communication. This makes the practical implementation of wireless communication in this band difficult. But backing up this wireless network with a wee structured optical network helps in solving this problem. This is what the Radio over Fiber (RoF) technology deals with. RoF is the technique by which the optical and wireless communication is seamlessly integrated. The optical fiber provides almost unlimited bandwidth and has no Optical fiber also has the immunity towards electromagnetic inference. Even though the initial cost for the implementation of the optical fiber network is high, the ease of maintenance makes the technology makes it one of the promising candidates for long haul communication.

The millimeter waves in optical domain can be easily transmitted between the central office and the remote station. The central office controls the entire signal processing functions and transmits signals to the remote station through the fiber network. The remote station only needs to convert the optical signals to electrical signals and transmit it wirelessly.

The generation of millimeter wave carrier is another important problem faced during the practical implementation of millimeter wave systems. As the frequency of operation is very high, the local oscillators which are to be designed must have high frequency capabilities. Also, the generation of millimeter waves in electrical domain is very difficult. The frequency limit of common electronic devices is the main reason for this. Hence optical generation is the feasible way[2]. Among many techniques used, optical heterodyning is an attractive method for generating optical carrier frequencies. This method can be called as Optical Frequency Multiplication (OFM). OFM is the process in which a low frequency RF is upconverted to a much higher microwave signal through optical signal processing. Frequency multiplication techniques based on non-linear modulation of Mach Zehnder modulator is used in this paper. This technique makes system more reliable and cheap to implement[3].

QAM being a popular vector digital modulation format is selected as the modulation scheme. Orthogonal Frequency Division Multiplexing (OFDM) technique is also used to provide high spectral efficiency and to achieve high level of spectral shaping[4]. OFDM allows the allocation of narrow guard bands to be reserved for each sides of the carrier. The OFDM signal in the EHF band is wirelessly transmitted from the remote station to the mobile units. wireless link is modeled as an AWGN channel for the analysis. The performance of this OFDM system is studied both in optical fiber and AWGN channel.

This paper is organized as follows. Section II describes the system model. The principle of operation of whole system is explained in section III. The details of the experiment done and the results are explained in section IV and finally the paper is concluded in section V.

II. SYSTEM MODEL

The whole system is modeled as shown in fig. 1. The transmission system is divided into three stages.

i. Central Office
ii. Remote station
iii. Mobile unit

The central office and the remote station are connected via an optical fiber link and the interconnection between the remote station and mobile station is wireless.

A. Central office

The central office is where all the signal processing functions are carried out. A random binary sequence is taken as the message signal. The binary sequence is fed to a quadrature amplitude modulator where the message bits are mapped to a 16-QAM signal. The 16-QAM signal is again modulated by an OFDM modulator. This OFDM modulated QAM signal serves as the RF signal to be transmitted over the fiber network. A continuous wave laser diode is used as the light source for the optical network. The OFDM signal is modulated on to the light waves with the help of electro absorption modulator. The modulation procedure is carefully designed so that the frequency up-conversion requirement is also satisfied. The optical signals carrying the millimeter waves are transmitted to the remote station through optical fiber.

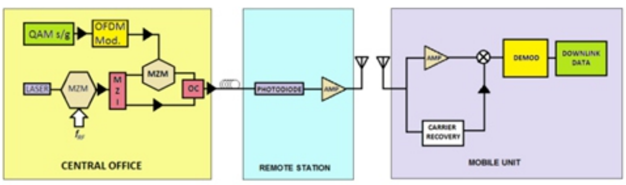

Fig. 1 System model

A. Remote Station

The remote station serves as the transmitter for the wireless communication link. The optical millimeter waves have to be converted to electrical signals for wireless transmission. As the power loss in the optical fiber link is low, the need for an RF amplifier can be avoided if sufficient power is by the central office itself. High speed photo diodes are for the conversion of optical millimeter wave to electrical signal and this signal is abstracted by using appropriate electrical filter. The RF millimeter wave is then fed to antenna for further transmission. The wireless channel between the remote station and mobile unit is modeled as an AWGN channel.

B. Mobile Unit

The mobile unit receives the millimeter wave signals with the help of in built antennas. As the operating frequency is in the order of Gigahertz, the size of the antenna required will be very small as antenna height and frequency are inversely proportional. The requirement for high frequency oscillators for demodulation can be avoided by the implementation of self homodyning or by extracting the carrier by narrow band filters and reusing the same. The OFDM and the QAM modulated signals are demodulated using demodulators and the input message sequence is obtained.

III. PRINCIPLES OF OPERATION

A. Optical Frequency Multiplication

In this technique, a low frequency RF is upconverted to a much higher microwave signal through optical signal processing. The mathematical formulation of the frequency

upconversion process is expressed as follows. In this paper we assume that the laser diode has a central frequency

$$f_c = 2\pi/\omega_c \qquad (1)$$

The light wave with central frequency f_c can be expressed as

$$E_c(t)=E_c exp(j\omega_c t) \qquad (2)$$

Using a Mach Zehnder Modulator (MZM), the optical carrier wave Ec(t) is intensity modulated in accordance with the RF carrier signal. Let the RF signal be expressed as

$$V_{RF}(t)= V_{RF} sin(2\pi f_{RF} t) \qquad (3)$$

The modulation process results in a spectrum where the the central carrier along with a number of side bands separated by a frequency gap of fRF. For obtaining frequency multiplication, the modulation index is enhanced in such way that the optical carrier and the first order side bands are completely suppressed. The output of the modulator should only have prominent positive and negative second order side bands. This can be obtained for a modulation index m= 2.405[2,3]. The output of the MZM which serves as the carrier for the RF message signal can be expressed as:

$$E_{OC}(t)=(\alpha/2)\, E_c exp(j\omega_c t)[exp\{j(\pi/V_\pi)\, V_{RF} sin(2\pi f_{RF}t)\}$$
$$+exp\{-j(\pi/V_\pi)\, V_{RF} sin(2\pi f_{RF}t)\}]$$
$$=\alpha\, E_c\, J_2(m)[\, expj(\omega_c\text{-}2\,\omega_{RF})t\text{-}\, expj(\omega_c+2\,\omega_{RF})t] \quad (4)$$

where α is the MZM insertion loss and J2 is the second order Bessel function. Also modulation index m is given by

$$m= \pi VRF/V\pi \qquad (5)$$

Now the two side bands are separated by a frequency gap of 4fRF. Using Mach Zehnder Interferometer or other filter structures, the positive and negative sidebands are separated and the positive side band is modulated with OFDM signal. The expressions for the same are as follows.

$$EOC\text{-}(t)= \alpha\ Ec\ J2(m)\ \{expj(\omega c\text{-}2\ \omega RF)t\} \qquad (6)$$

for the negative side band and

$$EOC+(t)= \alpha\ Ec\ J2(m)\ [expj(\omega c+2\ \omega RF)t\ +$$
$$(\pi/V\pi)\{S(t)\ expj(\omega c+2\ \omega RF + \omega IF)t\}\ + \qquad (7)$$
$$(\pi/V\pi)\{S(t)\ expj(\omega c+2\ \omega RF - \omega IF)t\}]$$

for the modulated positive sideband, where S(t)is the modulating signal. After aligning polarizations, the two signals
EOC-(t) and EOC+(t) are combined using optical coupler and transmitted. The transmitted signal ET(t) is given by:

$$E_T(t)=E_{OC-}(t)+E_{OC+}(t)$$
$$=\alpha\ E_c\ J_2(m)\ [expj(\omega_c+2\omega_{RF})t\ +\ expj(\omega_c\text{-}2\omega_{RF})t\ +$$
$$(\pi/V_\pi)\{S(t)\ expj(\omega_c+2\ \omega_{RF} + \omega_{IF})t\}\ +$$
$$(\pi/V_\pi)\{S(t)\ expj(\omega_c+2\ \omega_{RF} - \omega_{IF})t\}] \qquad (8)$$

At the remote station, with the help of a high speed PIN photo diode, the optical signal is converted to electrical signal. The photo current includes a dc, high frequency RF signal at 4ωRF + ωIF and 4ωRF - ωIF, the modulating signal at ωIF and carrier signal at 4ωRF. Either of the two millimeter wave signals at 4ωRF + ωIF or 4ωRF - ωIF can be used for wireless transmission, and the expression for the millimeter wave at 4ωRF - ωIF is obtained as:

$$I_{mmw}(t)=2\mu\alpha^2(\pi/V_\pi)\, E_c^2 J_2^2(m)[I(t)cos(4\omega_{RF}\text{-}\omega_{IF})t$$
$$+Q(t)sin(4\omega_{RF} - \omega_{IF})t] \qquad (9)$$

where I and Q are the in phase and quadrature components of the of the modulating signal respectively. From the equation, it can be inferred that the RF carrier frequency ωRF becomes four times the original value. Hence we can say that by supplying a low frequency signal, high frequency carrier can be optically generated by careful design.

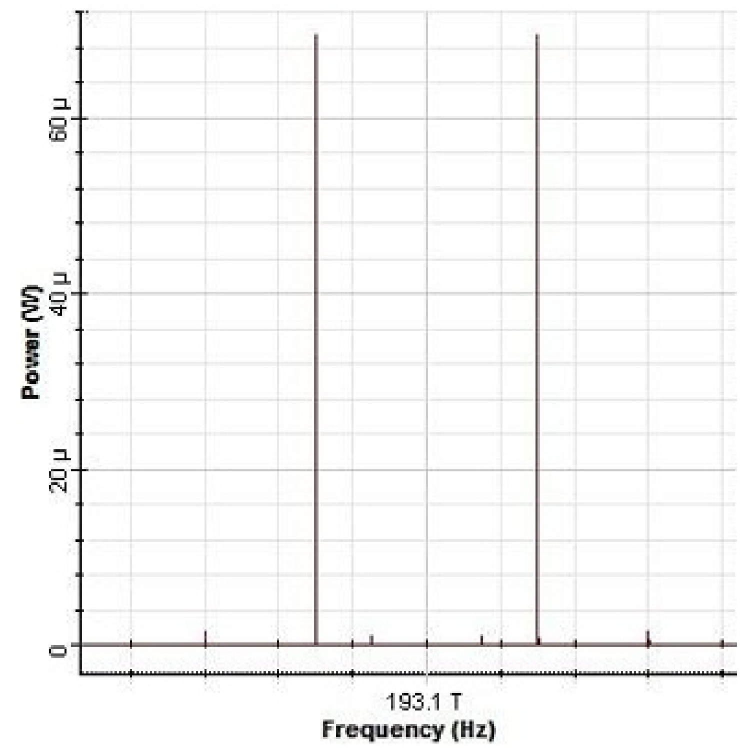

Fig.2 carrier suppression

B. RF Down Conversion

RF down conversion can be done using several methods. Two effective methods that can be used without the use of an external oscillator are self homodyning and carrier extraction and reuse method.

i. Self Homodyning

The method of extracting information encoded as modulation of an oscillating signal by comparing the same with a standard oscillation that would be identical to the signal itself is called homodyne detection. In optical interferometry, the reference radiation is derived from the same source as the signal before the demodulation process signifies homodyning[5].

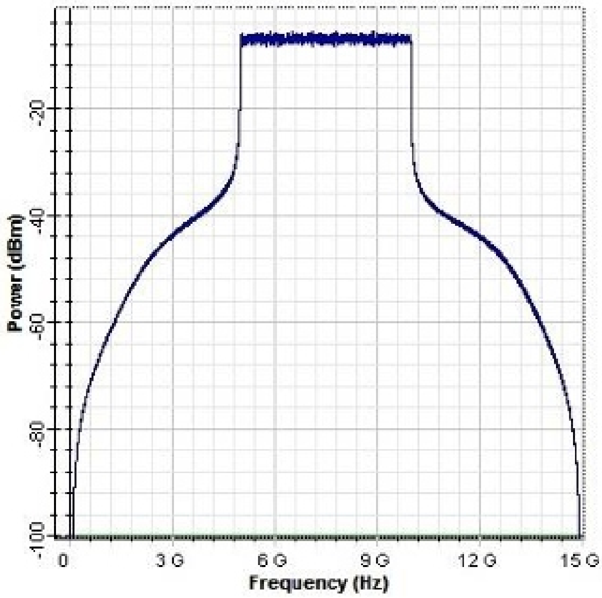

Fig.3 OFDM spectrum

ii. Carrier Extraction and Re-use

The carrier extraction and re-use is an RF technique where the high frequency RF carrier is extracted using narrow band filtering and reused by mixing it with the wide band data for down conversion. This procedure is carried out by first splitting the high frequency RF signal using a power splitter and at one of the power splitter's output, the RF carrier is filtered by narrow band filter. This filtered output serves as the local oscillator carrier signal which is mixed with high frequency received signal to the base-band signal.
These two methods provide cheap and efficient detection of the message signal at the receiver{5].

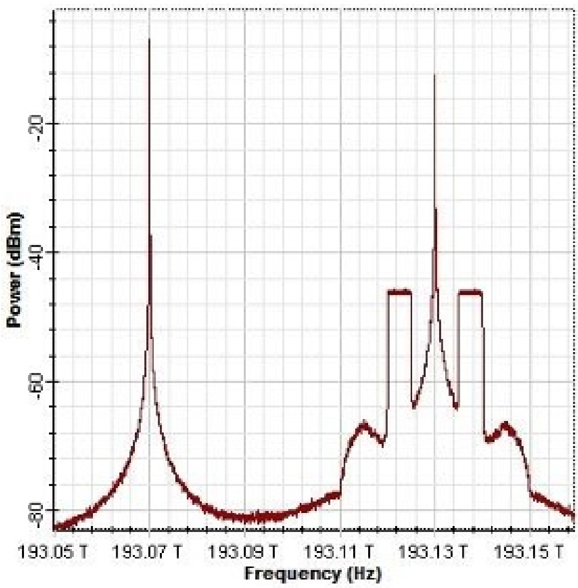

Fig. 4 Transmitted signal(spectrum)

IV. EXPERIMENTATION AND RESULTS

The optical link is built on opti-system platform. The central frequency of the continuous wave laser is selected as 193.1 THz. As the analysis in this paper is based on the V-band (40 – 75 Ghz), let's take the wireless link is operating at 60 GHz. For the generation of 60 GHz signal based on the OFM, the FRF is taken as 15 GHz. This 15 Ghz RF signal is fed to the MZM to obtain the two second order side bands at 193.13 THz and 193.07 THz. By enhancing the modulation index, the carrier as well as other side bands is suppressed completely as shown in fig.2. The Mach Zehnder Interferometer is used for splitting the two side bands and the upper side band is modulated with the OFDM signal.IV. EXPERIMENTATION AND RESULTS The optical link is built on opti-system platform. The central frequency of the continuous wave laser is selected as 193.1 THz. As the analysis in this paper is based on the V-band (40 – 75 Ghz), let's take the wireless link is operating at 60 GHz. For the generation of 60 GHz signal based on the OFM, the FRF is taken as 15 GHz. This 15 Ghz RF signal is fed to the MZM to obtain the two second order side bands at 193.13 THz and 193.07 THz. By enhancing the modulation index, the carrier as well as other side bands is suppressed completely as shown in fig.2. The Mach Zehnder Interferometer is used for splitting the two side bands and the upper side band is modulated with the OFDM signal.

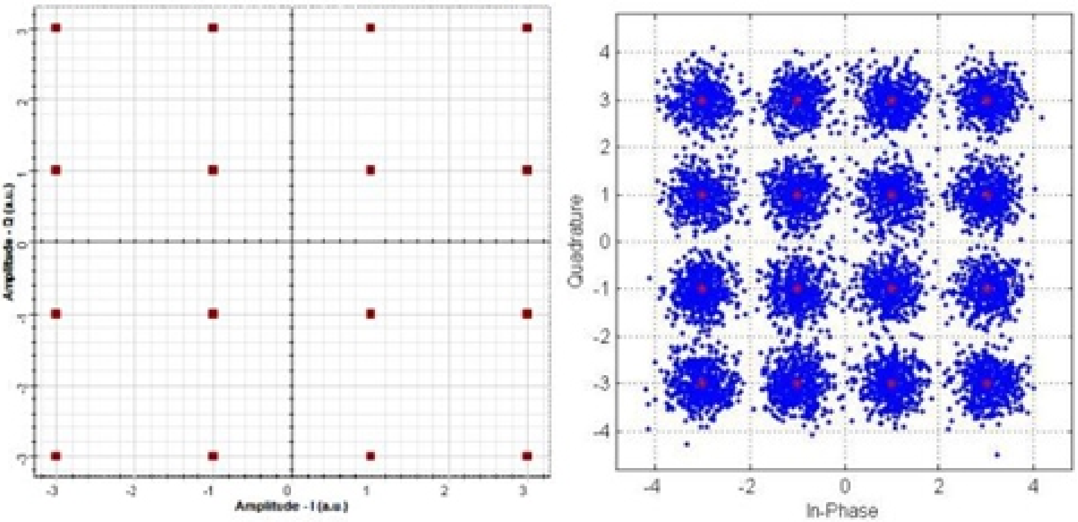

Fig. 5 constellation diagrams (transmitted and received)

A pseudo random binary sequence generator which produces bit sequences is taken as the information signal. The bit rate is taken to be 5 Gbps. The QAM modulator maps the bit sequence into a 16 QAM vector modulated signal. The in-phase and quadrature components are fed to an OFDM modulator having 512 sub-carriers. A block length of 1024 is taken for IFFT operation. 7.5 GHz intermediate frequency carrier is used to modulate the signal and the spectrum is shown in fig. 3. This OFDM modulated 7.5 GHz signal is modulated on to the positive second order side band at 193.13 THz and is then optically combined with the side band at 193.07 THz. The spectrum of the combined signal is in fig. 4.

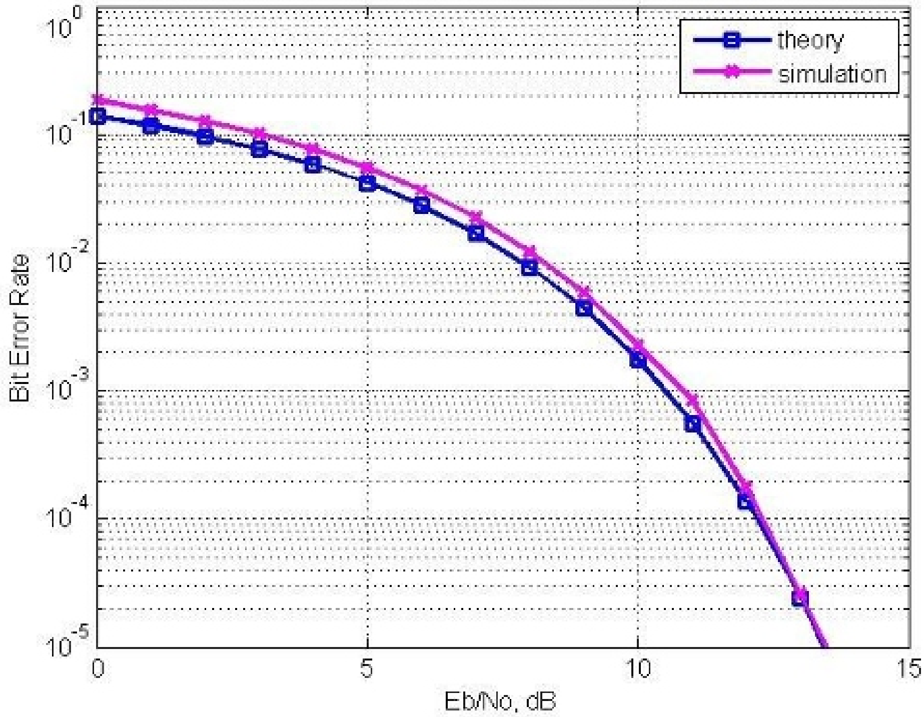

Fig. 6 BER versus SNR

The signal is transmitted to the remote station through standard single mode fiber with following standards. Chromatic dispersion = 17 ps/nmKm and power attenuation of 0.2 dB/Km. The photo diode converts the optical signals to millimeter wave electrical signals as shown in equation (9). The photo current consists of 60 GHz pure millimeter wave carrier, 67.5 GHz and 52.5 GHz OFDM modulated RF signals, 7.5 GHz original OFDM signal and dc component. In this simulation, the millimeter wave at 52.5 GHz and the millimeter carrier at 60 GHz are wirelessly transmitted. The channel for the wireless transmission is modeled as channel and is simulated by introducing MATLAB component in opti-system. Carrier extraction method is used for the down conversion and are demodulated to get the QAM signal back. The constellation diagram of both transmitted and received QAM signal is shown in fig 5.

The constellation diagram shows that the signal maintain sufficient distance between other signal points so that the probability of false detection remains low. The noise due to laser diode and photo diode also affects the system performance. Fig. 6 shows the BER plot of the received signal. It is seen that the bit error rates are below the forward error correction limit and also the theoretical and simulation results maintain satisfactory performance.

V. CONCLUSION

In this paper, the performance of both optical and wireless links in an OFDM based RoF system is studied. The optical frequency multiplication and the carrier extraction and reuse procedures provide cheap and reliable implementation of RoF networks and also maintain good performance not only in optical fiber but also in wireless AWGN channels. Carrier recovery and reuse in the mobile unit makes RoF systems simple and cost effective.

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BER Performance of Free-Space Optical System Over Gamma Gamma Turbulence with Pointing Error

Narendra Kumar Verma, Hemant Narayan

ABSTRACT

This paper investigates BER performance of free space optical (FSO) communication over gamma-gamma turbulence channel. Which is widely accepted model for moderate to strong atmospheric turbulence condition. By considering Atmospheric turbulence, Pointing error and Atmospheric attenuation a combined Statistical model for intensity fluctuation at the receiver is described for given weather and pointing error condition, a closed form expression is derived for BER performance of FSO communication System.

Keywords: Free-space optical communication, BER, Pointing error, Atmospheric turbulance., atmospheric loss

I. INTRODUCTION

Free Space Optical Communication is a line of sight (LOS) Technology. That transmit laser beam of light through atmosphere for a broad band communication [1] [2]. FSO system have its own advantage over existing RF system, FSO System is license free withy high bandwidth communication technology and FSO is prominent of last mile connectivity. It unique properties make it also appearing for a number of different application, including wide area network fiber backup, backhaul for wireless cellular network, Redundant link and disaster recovery[3] . Despite of this point to point laser signal is very secure for transmission. Having the lot of advantage of FSO Communication, various drawback should be taken into account in the desire of FSO link in the atmospheric turbulence ,which is result of variation refractive index to in homogeneity of air particle [4] .This degrade the performance of FSO system particular the FSO link which is greater than 1 km in length [5]. Another factor which degrade the performance of FSO link in building swag , which causes pointing error all a result of misalignment between the transmitter and receiver over FSO link[6]. Various statistical model have been proposed to describe atmospheric turbulence over different degree of atmospheric turbulence [77]. In particularly log normal is suitable model for only weak atmospheric condition and gamma-gamma have found to be suitable for moderate to strong turbulence condition respectively. Meanwhile as a standard performance metric adopted by most FSO system, there have been several study which described the performance parameter of the FSO communication system

II. SYSTEM AND CHANNEL MODEL

Consider a FSO link, the transmitter modulates data onto the instantaneous intensity of optical beam. The laser beam propagate through gamma-gamma turbulence channel the receive integrates the photocurrent signal which is related to incident power by the detector responsively for each bit period at the receiver the signal suffer from a fluctuation in signal intensity due to atmospheric turbulence and a misalignment as well as noise an can be modeled as

$$y = hRx + n \tag{1}$$

Where $x \in (0,1)$ and R is Responsivity of Photodetector ,h is normalized fading coefficient. Consider to be constant over a large no. of transmitted bits and n is AWGN with mean is zero and varience is σ_n^2.

Consider the channel fading coefficient h is modeled as using three phenomena, distance dependent atmospheric attenuation h_e ,Pointing error h_p and atmospheric turbulence h_a.the channel state is described as

$$h = h_e h_p h_a \tag{2}$$

In this Paper, we consider intensity modulated Direct detection (IM/DD) channel using on-off keying(OOK) modulation. The transmitted signal is equally probably from an OOK constellation such that $x \in (0,2p_t)$, and p_t is average transmitted power.

A. Atmospheric Attenuation:-

Atmospheric attenuation is a deterministic phenomenon which is best described by Beers-Lambert Law as[13]

$$h_e(z) = \exp(-\alpha z) \tag{3}$$

$h_e(z)$ is the atmospheric attenuation over a propagation distance in z and α is attenuation constant[3].

B. Atmospheric Turbulence

There has been significant research after finding accurate and efficient model for atmospheric turbulence. For weak turbulence log-normal is widely accepted model while for moderate to strong atmospheric turbulence Gamma-Gamma is a perfect distribution. And intensity Distribution is Gamma-Gamma Turbulance is given by[14] where Γ(・) denotes Gamma function [15, eq.(8.310.1)], Kv (・) is the v th-order modified Bessel function of the kind [15, eq.(8.432.2)], and α, β are the effective number of small-scale and large-scale eddies of scattering environment, respectively. According to [16], α and β can be obtained as

$$\alpha = \left[exp\left(\frac{-.49\sigma_r^2}{(1 + 1.11\sigma_r^{12/5})^{7/6}} \right) - 1 \right]^{-1} \tag{5}$$

$$\beta = \left[exp\left(\frac{-.51\sigma_r^2}{(1+.69\sigma_r^{12/5})^{5/6}} \right) - 1 \right]^{-1} \tag{6}$$

where σ_R^2 is the Rytov variance defined as [15]

$$\sigma_R^2 = 1.23 C_n^2 K^{7/6} L^{11/6} \tag{7}$$

where $k = 2\pi/\lambda$ is the optical number, λ is the wavelength, d is the propagation distance and $C2$ n (L) is the index of refraction structure parameter at altitude L [17].

C. Pointing Error Model

By considering a circular detection aperture of radius ρ and a Gaussian beam, the PDF of h_p can be derived using the assumptions and methodology described in [17] as

$$f_{h_p}(h_p) = \frac{\rho^2}{A_0^{\rho^2}} h_p{}^{\rho^2 - 1} \quad 0 \leq h_p \leq A_0 \tag{8}$$

The ratio of corresponding beam radius of the receiver to the pointing error displacement is denoted as $\rho = w_{zeq}/2\sigma_s$ and w_{zeq} is corresponding beam width with $w_{zeq}^2 = w_z^2\sqrt{\pi}\,\mathrm{erf}\,(v)/2\,v\,\exp\,(-v)^2$ and $A_0 = [\mathrm{erf}\,(v)]^2$ is the power received at r=0 .error function is denoted as erf(.) [5] (beam radius is calculated at e^{-2}) is referred as w_z beam waist and $v = \sqrt{\pi}r/\sqrt{2}\,w_z$.

D. Combined fading Model

Combined PDF for due to all these three fading parameter is given by following equation

$$f_h(h) = \int f_{h/_{h_l h_a}}\left({}^{h}/_{h_l h_a}\right) f_{h_a}(h_a)dh_a \tag{9}$$

Where $f_{h/_{h_l h_a}}\left({}^{h}/_{h_l h_a}\right)$ is the conditional probability given h_a state and given by

$$f_{h/_{h_l h_a}}\left({}^{h}/_{h_l h_a}\right) = \frac{1}{h_l h_a} f_{h_p}\left({}^{h}/_{h_l h_a}\right)$$

$$= \frac{\rho^2}{A^{\rho^2 h_l h_a}}\left(\frac{h}{h_l h_a}\right)^{\rho^2-1} \tag{10}$$

By substituting (4) and (10) into (9), the PDF of h is given

By

$$f_h(h) = \int_{h/A_0}^{\infty} \frac{\rho^2}{A^{\rho^2 h_l h_a}}\left(\frac{h}{h_l h_a}\right)^{\rho^2-1} \frac{2(\alpha\beta)^{\frac{\alpha+\beta}{2}}}{\Gamma(\alpha)\Gamma(\beta)} h_a^{\frac{\alpha+\beta}{2}} K_{\alpha-\beta}\left(2\sqrt{\alpha\beta I}\right) dh_a \tag{11}$$

According to [18, eq.(14)] and [18, eq.(26)], (11) can be rewritten as

$$f_h(h) = \frac{\alpha\beta\rho^2}{A_0 h_l \Gamma(\alpha)\Gamma(\beta)} \times G_{1\ \ 3}^{3\ \ 0}\left[\begin{matrix} & 1-\frac{\alpha+\beta}{2}+\rho^2 & \\ -\frac{\alpha+\beta}{2}+\rho^2 & \frac{\alpha-\beta}{2} & \frac{\beta-\alpha}{2}\end{matrix}\right](12)$$

Where $G_{p\ \ q}^{m\ \ n}[.]$ is the Meijer's G-function [15, eq.(9.301)].

Using [19, eq.(9.31.5)], (12) can further simplified as

$$f_h(h) = \frac{\alpha\beta\rho^2}{A_0 h_l \Gamma(\alpha)\Gamma(\beta)} *$$
$$G_{1\ \ 3}^{3\ \ 0}\left[\frac{\alpha\beta}{A_0 h_l}.h\middle|\begin{matrix} & \rho^2 & \\ \rho^2-1 & \alpha-1 & \beta-1\end{matrix}\right](13)$$

III. AVERAGE BER ANALYSIS

BER analysis plays a crucial role in FSO communication system. Mathematically, the BER of IM/DD with OOK can be given by P(e) = p (1) p (e /1)+p (0) p (e /0), where p (1) and

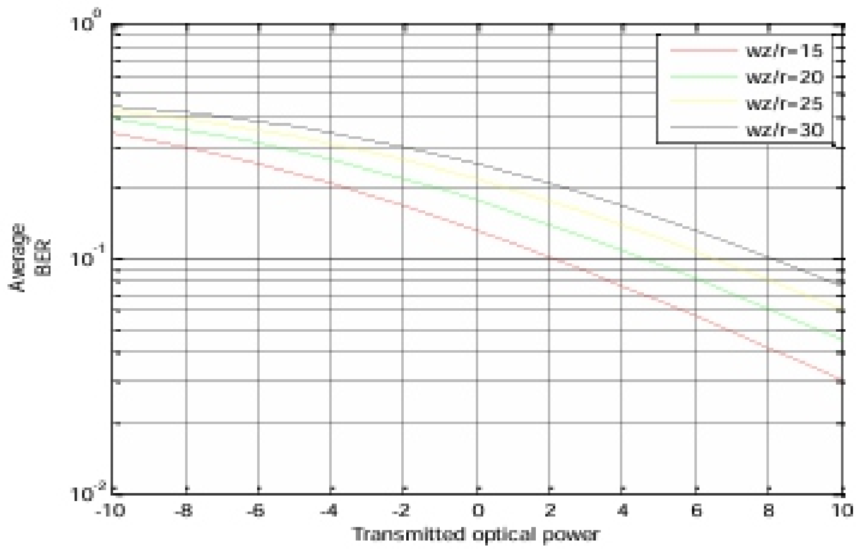
p (0) are the probabilities of sending 1 and 0 bits, respectively, and p (e /1) and p (e /0) denote the conditional bit error probabilities when the transmitted bit is 1 and 0, respectively. Assume that p (0) = p (1) = 1/2 and p (e /1) = p (e /0) , it is easy to show that conditioned on h, the bit error probabilities can be derived as follow

$$P\left({}^{e}/_{h}\right) = \frac{1}{2} erfc\left(\frac{P_t R h}{\sqrt{2}\sigma_n}\right) \quad (14)$$

$$P_b(e) = \int_0 P_b(e/h).\, f_h(h)\, dh \qquad (15)$$

Substituting (13) and (14) into (15), and using [20, eq. (06.27.26.0006.01)], [18, eq. (21)], [15, eq.(9.31.1)], a closed form expression for average BER is derived as

$$P(e) = \frac{2^{\alpha+\beta-4}\rho^2}{\sqrt{\pi^3}\Gamma(\alpha)\Gamma(\beta)} \times$$
$$G_{6\ 3}^{2\ 5}\left(\frac{8A_0^2 h_l^2 P_t^2 R^2}{\alpha^2\beta^2\sigma_n^2}\middle| \begin{matrix} \frac{2-\rho^2}{2}, \frac{1-\alpha}{2}, \frac{2-\alpha}{2}, \frac{1-\beta}{2} & \frac{2-\beta}{2} & 1 \\ 0 & \frac{1}{2} & \frac{-\rho^2}{2} \end{matrix}\right)$$



transmitted power for various values of the normalized beam width. It can be seen from Fig.1 that the average BER decreases with the increase of the transmitted signal power. Meanwhile, BER performance will be better if a narrow beam width is used. It is because the received signal power is increased when a narrow beam width is used. Moreover, it can be found that simulation results show close agreement with analytical results.

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Design of High Speed 5:2 Compressor for Fast Arithmetic Circuits

N. Srinivas, Y. Rajasree Rao

ABSTRACT

Multipliers are important components that dictate the overall arithmetic circuits' performance. The most critical components of multipliers are compressors. In this paper, a new 5:2 compressor architecture based on changing some internal equations is proposed. In addition, using an efficient full-adder (FA) block is considered to have a high-speed compressor. The proposed architecture is compared with the best existing designs presented in the state-of the-art literature in terms of power, delay and area.

Keywords: Full-Adder (FA), XOR-XNOR, Multiplexers.

I. INTRODUCTION

Multipliers are one of the most significant blocks in computer arithmetic and are generally used in different digital signal processors. There is growing demands for high speed multipliers in different applications of computing systems, such as computer graphics, scientific calculation and image processing and so on. Speed of multiplier determines how fast the processors will run and designers are now more focused on high speed with low power consumption. The multiplier architecture consists of a partial product generation stage, partial product reduction stage and the final addition stage. The partial product reduction stage is responsible for a significant portion of the total multiplication delay, power and area. Therefore in order to accumulate partial products, compressors usually implement this stage because they contribute to the reduction of the partial products and also contribute to reduce the critical path which is important to maintain the circuit's performance. Most computerized mathematic applications are executed utilizing digital logic circuits, in this manner working with a high degree of reliability and precision. In any case, numerous applications, for example, in multimedia and image processing can endure mistakes and imprecision in calculation and still produce important and helpful results. Exact and precise models and algorithm are not generally suitable or productive for use in these applications. The paradigm of inaccurate calculation depends on relaxing completely precise and totally deterministic building modules when for instance, planning energy efficient system. This permits uncertain calculation to divert the current design procedure of computerized circuits and systems by exploiting a reduction in multifaceted nature and expense with conceivably a potential increment in execution and force productivity.

In exact (or vague) figuring depends on utilizing this property to design disentangled, yet inexact circuits working at higher execution and/or lower power utilization contrasted and exact (definite) logic circuits. Expansion and multiplication are broadly utilized operations as a part of computerized mathematic; for expansion adders and proposed a few new measurements for assessing rough and probabilistic adders as for brought together figures of legitimacy for outline appraisal for estimated processing applications. For every data to a circuit, the Error Distance(ED) is characterized as the mathematic separation between a mistaken yield and the right one. The Mean Error Distance (MED) and normalized error distance (NED) are proposed by considering the averaging impact of various inputs and the standardization of multiple-bit adders. The NED is almost invariant with the measure of an execution and is accordingly valuable in the unwavering quality evaluation of a particular configuration. The tradeoff in the middle of accuracy and force has additionally been quantitatively assessed in.

II. COMPRESSOR ARCHITECTURES

Compressors are building blocks used for accumulating partial products during the multiplication process. The basic idea in an n : 2 compressor is that n operands can be reduced to two, by doing the addition while keeping the carries and sums separate. This means that all of the columns can be added in parallel without relying on the result of the previous column, creating a two-output adder with a time delay that is independent of the size of its inputs. The full adder is the most primitive compressor and is often referred to as the 3 : 2 compressor since it compresses three operands into two. The sum and carry outputs are given by the following set of equations:

$$\mathrm{Sum} = x_1 \oplus x_2 \oplus c_i$$
$$\mathrm{Carry} = x_1x_2 + x_2c_i + x_1c_i$$

Where x1 and x2 are the input operands and ci is the carry from previous stage. This 3:2 compressor has a delay of two XOR gates, and is normally used in carry-save form to sum up the partial products in a multiplier tree. Though the addition used in this manner is much faster than that of a ripple carry adder, the interconnections are very irregular thereby making the structure more complex. The 4:2 compressor was initially designed by an connection of two 3:2 compressors as shown in Fig 1. The structure has a delay of four XORs. The advantage of the structure lies its carry free nature, whereby the carry from the previous stage is not propagated to the next stage.

A novel design of a 4:2 compressor with XORs and multiplexers (MUX) as building blocks is presented in [5]. This design is based on a modified set of equations for the sum and carry outputs as:

$$\mathrm{Sum} = x_1 \oplus x_2 \oplus x_3 \oplus x_4 \oplus c_i$$

$$\mathrm{Carry} = (x_1 \oplus x_2 \oplus x_3 \oplus x_4)c_i + (\overline{x_1 \oplus x_2 \oplus x_3 \oplus x_4})x_4,$$

$$C_o = (x_1 \oplus x_2)x_3 + (\overline{x_1 \oplus x_2})x_1$$

A widely used compressor of significant importance is the 5:2 compressor. Its block diagram is shown in Fig. 2. It has seven inputs of which five are direct inputs and two are carry-in bits from a previous stage. Similarly, there are four outputs of which two are carry-out bits to the next stage and the other two are sum and carry bits. All the 5:2 compressors of different designs abide by the generic equation:

$$x_1 + x_2 + x_3 + x_4 + x_5 + c_{i1} + c_{i2} = \mathrm{Sum} + 2(\mathrm{Carry} + C_{o1} + C_{o2})$$

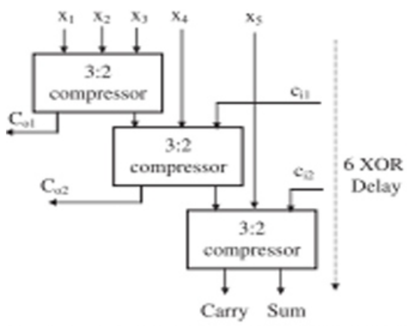

Fig 1: 5:2 Compressor Implementation using 3:2 compressors

cascading three 3:2 compressors as shown in Fig 1. This structure has a delay of 6 XORs and is slower than the 6:2 compressor presented in [3], which has a delay of only five XORs. A faster implementation of the 5:2 compressor with 5 XOR delays is presented in [6].

III. PROPOSED OPTIMIZED 5:2 COMPRESSOR DESIGN

In this section the proposed architecture is introduced. Two improvement approaches are used to propose the new 5:2 compressor architecture. First, by a closer look at dashed box of Fig. 1, it represents the functionality of a conventional FA and can be replaced by variety of Fas presented in the literature. This replacement is expected to lead to considerable speed improvement, due to 34% faster operation of CMOS FA in comparison to two cascaded CMOS XOR gates, as it is explained in [14]. Therefore, the CMOS FA presented in [15] is used in our proposed 5:2 compressor architecture. To further improvement, we make some changes to internal equations of the 5:2 compressor to eliminate final Not gates of the CMOS FA. By doing so, we could have reduced power dissipation as well as improved operational speed. To achieve this goal, we have to use XNOR gates instead of XORs of the second stage of the architecture. Hence, we propose the architecture which is shown in Fig 2

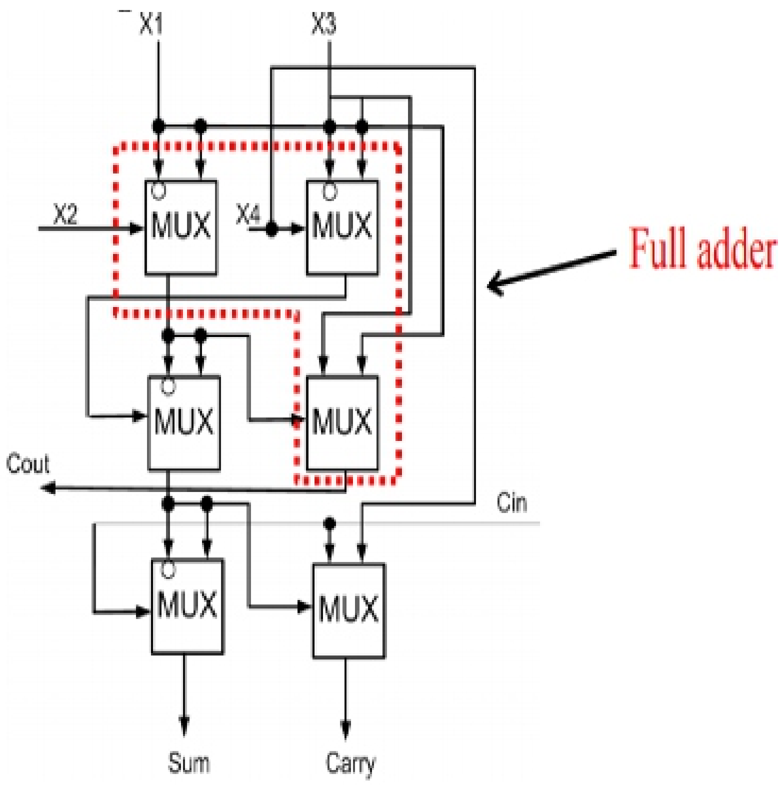

Fig 2: optimized design of 5:2 compressor

$$Sum = A \oplus B \oplus C = ABC + \boxed{\bar{A}\bar{B}C + A\bar{B}\bar{C} + \bar{A}B\bar{C}}$$

$$Carry = AB + BC + AC$$

$$\overline{Carry} = \boxed{\bar{A}\bar{B} + \bar{B}\bar{C} + \bar{A}\bar{C}}$$

By taking the NOT of Carry, we could use part of the circuit, which generates Sum signal, to generate Carry signal. Thus the higher performance of full adder could be achieved. Based on the above formulae, it is conjectured that lowering the critical path delay of the (5:2) compressor. However, it is very likely to explore different logic design styles at transistor level to achieve significantly improved low power and high-speed (52) compressor. However, in terms of power dissipation, Design 1 might be a better choice due to its use of MUX for implementing Co1, even though both circuits may have comparable values for power. In this section, the impact of using the proposed compressors for multiplication is investigated. A fast (exact) multiplier is usually composed of three parts (or modules). In the design of a multiplier, the second module plays a pivotal role in terms of delay, power consumption and circuit complexity. Compressors have been widely used [9, 10] to speed up the CSA tree and decrease its power dissipation, so to achieve fast and low-power operation. The use of approximate compressors in the CSA tree of a multiplier results in an approximate multiplier.

IV. SIMULATION RESULTS

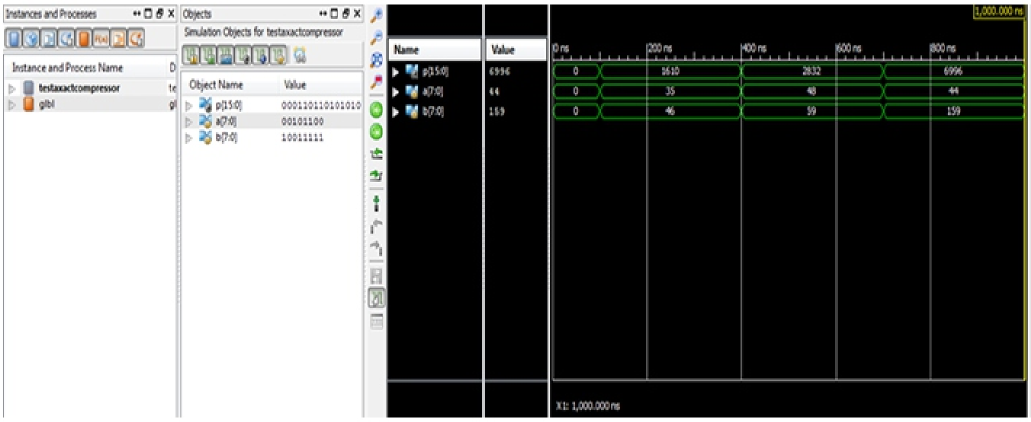

Fig 3: simulation result for the proposed dadda multiplier using optimized 5:2 compressor

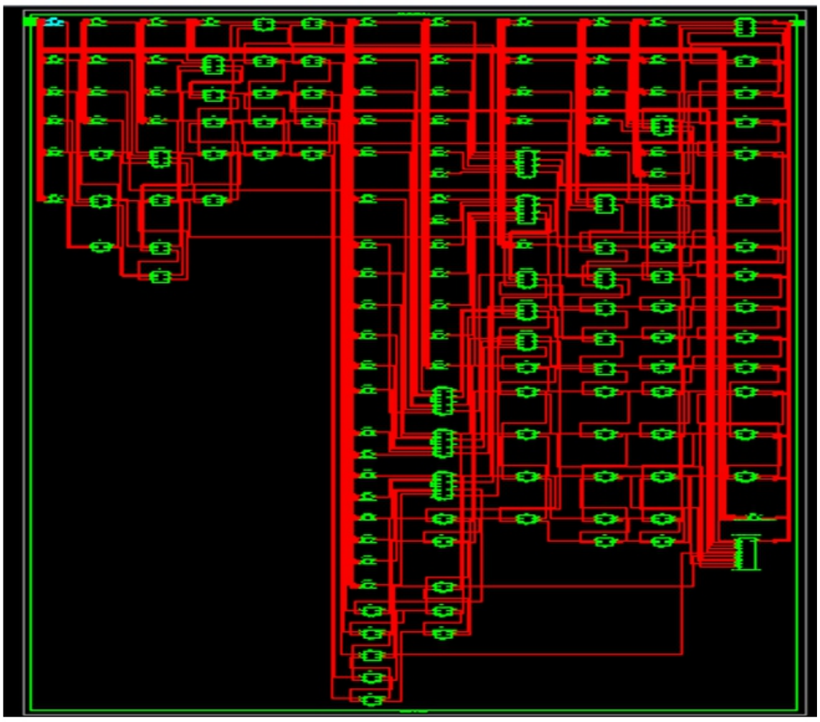

Fig 4: RTL of the proposed dadda multiplier using optimized 5:2 compressor

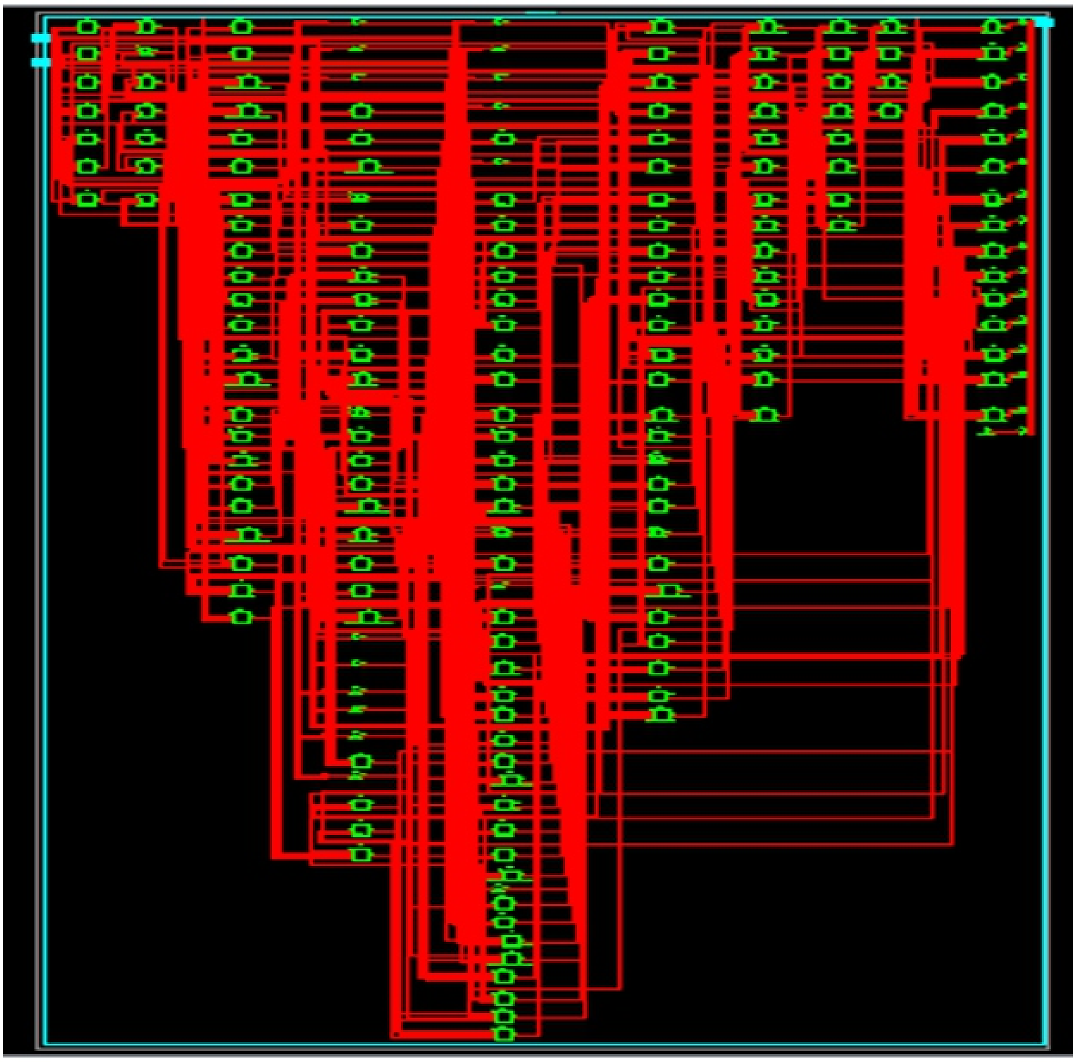

Fig 5: Technology schematic of the proposed dadda multiplier using optimized 5:2 compressor

exactmultiplication Project Status (06/28/2017 - 00:12:44)			
Project File:	jkl.xise	**Parser Errors:**	No Errors
Module Name:	exactmultiplication	**Implementation State:**	Synthesized
Target Device:	xa3s400-4pqg208	• **Errors:**	No Errors
Product Version:	ISE 14.7	• **Warnings:**	2 Warnings (0 new, 0 filtered)
Design Goal:	Balanced	• **Routing Results:**	
Design Strategy:	Xilinx Default (unlocked)	• **Timing Constraints:**	
Environment:	System Settings	• **Final Timing Score:**	

Device Utilization Summary (estimated values)			[-]
Logic Utilization	**Used**	**Available**	**Utilization**
Number of Slices	91	3584	2%
Number of 4 input LUTs	160	7168	2%
Number of bonded IOBs	32	141	22%

Fig 6: summary report for the proposed dadda multiplier using optimized 5:2 compressor

V. CONCLUSION

In this paper we examined the various 5:2 compressors available in the literature and then proposed two new highspeed 5:2 compressor architectures. The new designs limit the carry propagation delay to a single compressor stage. They are then analysed for their critical delay paths and are found to be faster than the best one reported in the literature. It is concluded that the proposed architecture consumes the lowest power, operates faster and consequently is the most energy efficient one.

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