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CONTENTS

BHARATI VIDYAPEETH POONA PUNE

Bharati Vidyapeeth's
Institute of Computer Applications and Management
A-4, Paschim Vihar, Rohtak Road, New Delhi-63

Email : bijit@bvicam.ac.in, Website : http://www.bvicam.ac.in

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Address for Correspondence:
Prof. M. N. Hoda
Editor-in-Chief, BIJIT
Director, Bharati Vidyapeeth's Institute of Computer Applications and Management (BVICAM),
A-4, Paschim Vihar, Rohtak Road, New Delhi – 110063 (INDIA). Tel.: +91 – 11 – 25275055
Fax: +91 – 11 – 25255056; E-Mail: bijit@bvicam.ac.in, Visit us at www.bvicam.ac.in/bijit

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Editorial Board

Editorial

It is a matter of both honor and pleasure for us to put forth the sixteenth issue of BIJIT; the BVICAM's International Journal of Information Technology. It presents a compilation of eleven papers that span a broad variety of research topics in various emerging areas of Information Technology and Computer Science. Some application oriented papers, having novelty in application, have also been included in this issue, hoping that usage of these would further enrich the knowledge base and facilitate the overall economic growth.

As a matter of policy of the Journal, all the manuscripts received and considered for the Journal, by the editorial board, are double blind peer reviewed independently by at-least two referees. Our panel of expert referees posses a sound academic background and have a rich publication record in various prestigious journals representing Universities, Research Laboratories and other institutions of repute, which, we intend to further augment from time to time. Finalizing the constitution of the panel of referees, for double blind peer review(s) of the considered manuscripts, was a painstaking process, but it helped us to ensure that the best of the considered manuscripts are showcased and that too after undergoing multiple cycles of review, as required.

The eleven papers, that were finally published, were chosen out of seventy three papers that we received from all over the world for this issue. We understand that the confirmation of final acceptance, to the authors / contributors, sometime is delayed, but we also hope that you concur with us in the fact that quality review is a time taking process and is further delayed if the reviewers are senior researchers in their respective fields and hence, are hard pressed for time.

*We further take pride in informing our authors, contributors, subscribers and reviewers that the journal has been indexed with some of the world's leading indexing / bibliographic agencies like **INSPEC** of IET (UK) formerly IEE (UK), **Index Copernicus International** (Poland) with **IC Value 7.76** for 2014, **ProQuest** (UK), **EBSCO** (USA), **Open J-Gate** (USA), **DOAJ** (Sweden), **Google Scholar, WorldCat** (USA), **Cabell's Directory** of Computer Science and Business Information System (USA), **Academic Journals Database**, **Open Science Directory, Indian Citation Index, etc.** and listed in the libraries of the world's leading Universities like **Stanford University, Florida Institute of Technology, University of South Australia, University of Zurich**, etc. Related links are available at http://www.bvicam.ac.in/bijit/indexing.asp. By now, BIJIT has 357* *citations, 09 as h-index and 08 as i10-index to its credentials. It is a matter of great satisfaction that BIJIT is enlisted at Sr. No. 6137 released by the* ***University Grants Commission (UGC),*** *New Delhi for the purpose of adjudicating the weightage of the published papers for Career Advancement Scheme (CAS) and Direct Recruitment of Teachers and other academic staff, as required under the UGC (Minimum Qualifications for Appointment of Teachers and other Academic Staff in Universities and Colleges). These encouraging results will certainly further increase the citations of the papers published in this journal thereby enhancing the overall research impact.*

*We take pride in informing our authors that Vol. 09 No. 01 i.e. from January, 2017 onwards; BIJIT shall be published by Springer, one of the world's largest publishers of scientific documents, under the title "****International Journal of Information Technology****" [An official Journal of Bharati Vidyapeeth's Institute of Computer Applications and Management (BVICAM), New Delhi] with acronym* ***BJIT*** *having* ***ISSN 2511-2104*** *(for Print Version) and* ***ISSN 2511-2112*** *(for Electronic Version), quarterly. The copy of the newly designed title page of the journal named as '****BJIT****' is printed at the back cover of this issue and further details, regarding the same, are available at the Springer Website at http://www.springer.com/computer/journal/41870. This is, certainly, a major achievement towards our commitment in realizing our vision "to achieve a standard comparable to the best in the field and finally become a symbol of quality".*

We wish to express our sincere gratitude to our panel of experts in steering the considered manuscripts through multiple cycles of review and bringing out the best from the contributing authors. We thank our esteemed authors for having shown confidence in BIJIT and considering it a platform to showcase and share their original research work. We would also wish to thank the authors whose papers were not published in this issue of the Journal, probably because of the minor shortcomings. However, we would like to encourage them to actively contribute for the forthcoming issues.

The undertaken Quality Assurance Process involved a series of well defined activities that, we hope, went a long way in ensuring the quality of the publication. Still, there is always a scope for improvement, and so, we request the contributors and readers to kindly mail us their criticism, suggestions and feedback at bijit@bvicam.ac.in and help us in further enhancing the quality of forthcoming issues.

Editors

CONTENTS

BIJIT - BVICAM's International Journal of Information Technology
Bharati Vidyapeeth's Institute of Computer Applications and Management (BVICAM), New Delhi (INDIA)

Measuring the Craniofacial Growth for Determination of Human Age through Classifiers

Sreejit Panicker[1], Smita Selot[2] and **Manisha Sharma[3]**

Submitted in Oct, 2015; Accepted in July, 2016

Abstract – An individual face reveals an array of information that can be age, gender or identity. The features play an important task in the estimation of age for a given person, just by looking at the face. In this research, the work is to design a model which classifies age with respect to features taken out from individual facial images by means of Neural Network (NN). In recent years, Artificial Neural Network (ANN) has been extensively used as a means for solving many decision making problems. In this paper for classifying age group a feed forward propagation neural networks is constructed from gray-scale facial images. Age groups are classified in four groups, including babies, young, middle-aged, and adults, applicable in the classification method. The course of action of the system is partitioned into three segments: locality, feature extraction, and age classification. The global features used to distinguish child from middle aged and adults is based on the ratios computed using the eyes, nose, mouth, chin, virtual-top of the head and the sides of the face as those features.

Index Terms – Age determination, Facial Feature parameter, Neural Networks.

1.0 INTRODUCTION

Aging and determining the age of human through facial parameters using various approaches in the area of computer vision and pattern recognition have gained considerable importance in recent times. With the advancement in technology, age determination is need for various real world applications. Aging is a natural phenomenon which exhibits the changes, more evident in context to facial growth.

Global features are considered by many researchers in varied areas such as identification, classifying gender, expressions and so on. Considering Age estimation and classification with facial features there is scope of further improvement. Researchers have worked for age estimation which give results for ages, or classify the ages in categories such as child, middle aged and old using extracted features.

An appropriate approach for age estimation to classify age in specific range is still a demanding problem. Thus, we focus the study on more specific extraction of facial features using statistical methods.

To achieve our goal, we create a database with extracted characteristics that is used to test and train the proposed approach; also a proper ANN model is built to address the problem. Age estimation through machine intelligence is a complex and demanding task. An individual is different in terms of aging, which cannot be known by his gene, but other factors also contributes such as the fitness, living approach, working style and sociality. Different ages have different forms of aging. The shape change (craniofacial growth) is visible in early years to teens; gradually the size of the face gets larger in later years. The major growth change noticeable is skin aging (texture change) which happens while aging from youth to adulthood.

The change in shape is a continuous process, but as the age increases its not significant. Therefore, facial aging is unruly and adapted. Males and females aging patterns are different, mainly because of cosmetics used by females that are likely to show younger appearances.

The paper is planned in the following manner: Section II discusses contribution from researchers in terms of feature extraction and classification problem. Section III elaborates the approach proposed for feature extraction by applying statistical methods Section IV illustrates the experimental results on applying the method. Section V provides the conclusion of the result.

2.0 RELATED WORK

Yen et al. [1] proposed a scheme based on allocation of values generated with the edge density of the given image. In the initial stage a face is anticipated to an ellipse, and genetic algorithm is applied to look for the best ellipse region to go with. In the subsequent extraction of feature, genetic algorithm is applied to the predefined sub regions such as eyes, nose and mouth.

Ramesha et al.[2] proposed age classification algorithm with extracted features using small training sets which gives improved results even if one image per person is available. It is a three stage process which includes preprocessing, feature extraction and classification. The facial features are identified using canny edge operator for detecting facial parts for extraction of features, and are subjected to classification using Artificial Neural Network.

Gu et al.[3] proposed automatic extraction of feature points from faces. A possible approach to find the eyeballs, close to and distant corners of eyes, center of nostril, and corners of mouth was adopted. Suo et al.[4] represented a compositional model using hierarchical And – Or graph that shows face in a

[1]Dept. of Computer Applications, Shri Shankaracharya Technical Campus, Chhattisgarh, INDIA.
[2]Dept. of Computer Applications, SSTC, Junwani, Bhilai, Chhattisgarh, INDIA.
3 Dept. of Electronics& Telecommunications, Bhilai Institute of Technology, Chhattisgarh, INDIA.
Email: [1]sreejit.bhilai@gmail.com

particular age group. In this method the And nodes disintegrate a face into parts to reveal details (e.g., hair, wrinkles, etc.) crucial for age perception and Or nodes represent variety of faces by applying different selection. Quantitative statistical analysis validates the performance of aging model and age estimation algorithm.

Ramanathan et al.[5] developed a transformation model that formulates the shape as a physical parametric muscle model that confines the delicate changes facial features go through with growing age.

Andreas Lanitis et al.[6] generated a model of facial appearance that uses statistical methods. It was further used as the source for generating a set of parametric depiction of face images. Based on the model classifiers were generated that accepted the form of representation given for the image and generate an approximation of the age for the face image. With the given training set, based on different clusters of images, classifiers for every age group were used to estimate age. Thus as given requirement in terms of age range the most appropriate classifier was selected so as to compute accurate age estimation.

3.0 OVERVIEW OF OUR APPROACH

In our approach to facial feature extraction we select the input image and crop the image, the cropped image is then normalized in shape and texture for further processing.

After these pre processing done to the input image we compute the mean value within the cropped image area and the number and area in pixels of the normalized image. The normalized image is then applied for feature extraction by using facial parameters.

The facial model in our approach Geometric Facial Measurement Model (GFMM) has various landmark points which comprise the feature set for further analysis using ANN classifiers as shown in figure 1. The facial model with parameters is revealed in figure 2.

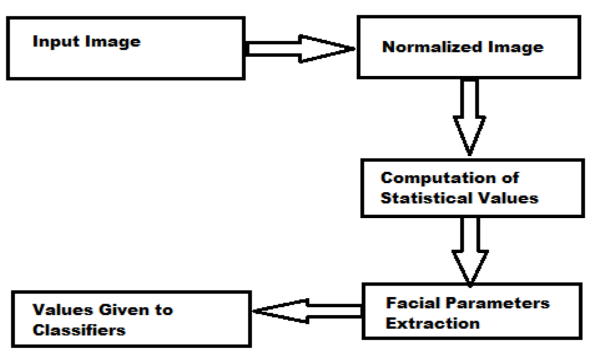

Figure 1. Diagram for performing GFMM

The details of each feature ID in the figure 1 is elaborated in table 1.

These facial parameters are used to measure the distance between the given points for the various subjects in our FGNET facial aging database. The computed values are then organized in different groups for classification of age.

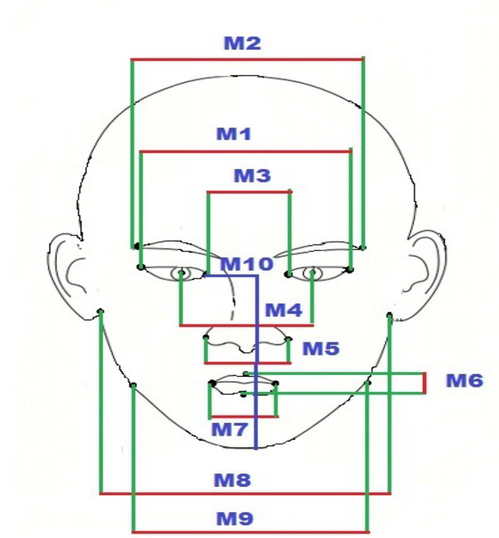

Figure 2. Facial Feature parameters

Table 1. Illustrates Facial Features in the Model

Feature Id	Feature parameter
M1	Extreme ends of left and right eye
M2	Extreme ends of left and right eyebrow
M3	Left and right eye points between nose
M4	Between left and right iris
M5	Nose end points
M6	Lips vertical measurement
M7	Lips horizontal measurement
M8	Ear points left and right
M9	Cheek points left to right
M10	Vertical measurement from nose

3.1 Mathematical Formulations

After we cropped the image it is subjected to normalization which is further processed to estimate the area. The value is a scalar that corresponds to the entire pixel number in the normalized image, at times it may not be the same because pixels with varied patterns are weigh differently. We use these values to compute Mean, each row or column of the input with the vectors of a particular dimension of the input, or complete.

4.0 EXPERIMENTAL RESULTS

4.1 Database for Aging

The Face and Gesture Recognition Research Network (FG-NET) is a database of face images of persons at their different ages. FG-NET is widely preferred for age related research works, because it contains 1,002 images of high resolution color or gray scale for performing various tasks. The age of persons in database varies from 0 to 69 years in chronological order of their aging. It comprises of 82 multiple race images with difference of lighting, pose and different expressions. The main effort to develop such an database was to help the researchers who perform various operations on facial image to study the aging effects. The database is available for free access for research purpose.

The approach used is implemented to FGNET aging database. The GFMM is a graphical based implementation for feature extraction from input image. The original input image and normalized image is shown in figure 3.

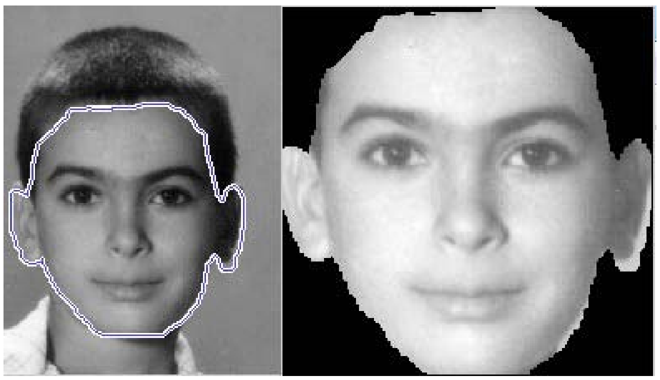

Figure3. Cropped and Normalized Image for Processing.

The normalized image is subjected for feature extraction here the distance between the points given in the feature ID is selected. After plotting all the facial feature parameters the values are computed for age classification problem. The figure 4 shows the facial feature parameters with their values.

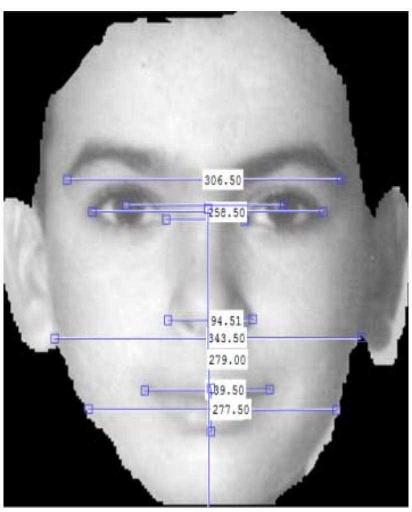

Figure4. Facial Feature Extraction.

The computed values are plotted for further analysis of the feature which is considered in different age groups. Broadly four groups are identified in which different images from FGNET database are subjected to further classification. The difference in values computed are evident from the graphs plotted against the values of figure 4 and 5 as shown in figure 6 and 7 respectively.

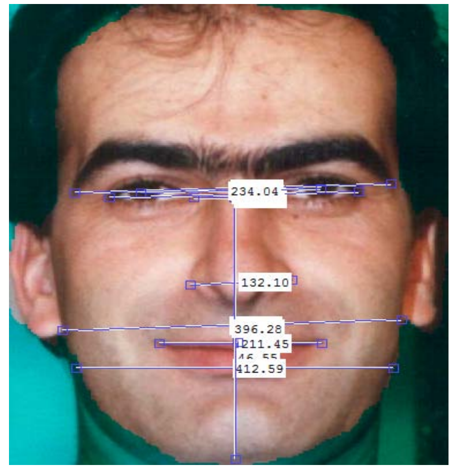

Figure5. Facial feature extraction by GFMM

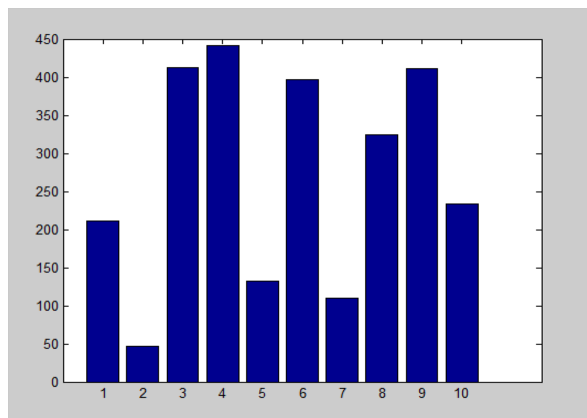

Figure6. Graph plotted against figure 4 values.

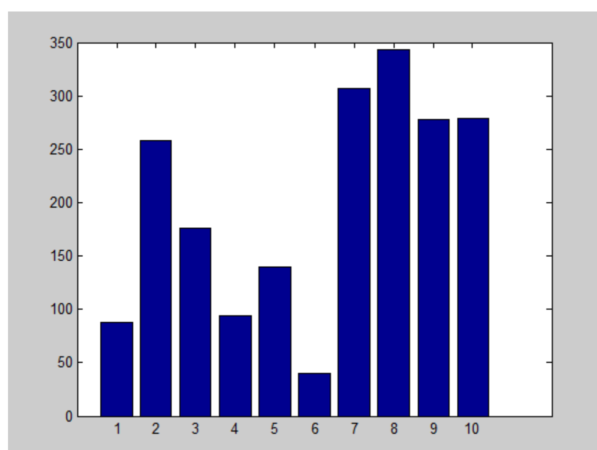

Figure7. Graph Plotted against figure 5 values.

4.2 Training with FGNET Dataset

In training we have used 101 images of different subjects with varying ages, in all age groups. We have classified four groups child, young, middle aged and old aged. The results of the training the input with its computational efficiency is made known in table 2

Table 2. Outcome on FGNET Database.

Performance parameter	Values (%)
accuracy	82%
sensitivity	84%
specificity	81%
Recall	74%
precision	84%
f_measure	85%
gmean	82%

5.0 CONCLUSION AND FUTURE SCOPE

The proposed approach is based on facial features for age estimation using Geometric Facial Measurement Model (GFMM). Our proposed method of finding facial features is different from other researchers. We have considered different facial parameter points through which we computed the statistical values. Normally the facial feature depends on 68 landmark points taken from facial images [7]. To test our system, we used FGNET aging database. A comparative learning has been carried out between different age group to assess the output of our projected system. It is evident from the training output that the proposed system performs closer to the human's judgment to identify age. In the testing phase, images are classified, while classification identification rate of the method is 85.6, on comparing this result with the human perception to age it exhibits a minor deviation to actual. It is evident from the study that performance of the system is near to values that is available in the database of aging used for classification.

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An Efficient Speaker Recognition System by Employing BWT and ELM

A. Indumathi[1] and **E Chandra**[2]

Submitted in April, 2016; Accepted in July, 2016

Abstract – Speaker recognition system has gained substantial research interest, owing to security enforcement in many applications. Mostly, the speaker recognition system is employed for achieving access control. Hence, a speaker recognition system must be capable of achieving greater recognition accuracy rates irrespective of the noise presence. This paper presents a novel speaker recognition system which tends to suppress the noise before the process of feature extraction. This idea improves the recognition accuracy of the system. Additionally, the proposed work can manage the noise mismatch between both the training and testing phase. Bionic Wavelet Transform (BWT) is applied to suppress the noise and the cleansed signal is obtained. This is followed by extracting Mean Hilbert envelope Coefficients (MHEC) and Power Normalized Cepstral Coefficients from the cleansed signal. Finally, the classification is achieved by Extreme Learning Machine (ELM). From the performance evaluation, it is evident that the proposed work shows convincing recognition accuracy rates, Signal to Noise ratio (SNR) and Mean Square Error (MSE).

Index Terms – Speaker recognition system, noise suppression, feature extraction, classification.

NOMENCLATURE

BWT-Bionic Wavelet Transform,*MHEC*-Mean Hilbert envelope Coefficients, ELM- Extreme Learning Machine, SNR-Signal to Noise ratio, MSE-Mean Square Error, MFCC -Mel Frequency Cepstral Coefficients, PLP-Perceptual Linear Prediction

1.0 INTRODUCTION

The process of communication relies on two entities namely speaker and listener [1]. On the whole, a message is transferred by means of words. However on keen observation, the language of the speech, emotion and the identity of speaker can be gained. The objective of a speaker recognition system is to extract features from the speech signal and to figure out the identity of the speaker [2]. Generally, a speaker recognition system involves two important processes, which are speaker verification and identification [3,4]. Speaker verification verifies the identity of the speaker by utilizing a sample speech. Speaker identification intends to detect the speaker from a set of speech samples.

[1,] Dept.of Computer Science, Dr SNS RCAS
[2]Bharathiar University, Coimbatore, Tamil Nadu
[1] endhumathi@gmail.com and [2] crcspeech@gmail.com.

Speaker recognition system is mostly employed for security based applications such as access control, authentication, law enforcement and personalization [5].Many such speaker recognition systems require clean speech signal for proving high recognition accuracy have been presented. However, the issue being caused by noise is not addressed in most of the works.Most of the real time applications suffer fromexternal noise,which has a serious impact over the quality of the system [6,7].

The features being extracted from the noisy speech will turn dissimilar to that of the training dataset. This paves way for higher misclassification rates. Besides this, many existing works provide solutions for noise removal [8]; however, it is not mandatory that the training and testing samples of speech should possess the same type of noise.

This paper strives to present a novel speaker recognition system that overthrows the noise by means of bionic wavelet transform. The features are then extracted from the cleansed speech signal and finally classified. The Mean Hilbert envelope Coefficients (MHEC) and Power Normalized Cepstral Coefficients (PNCC) are extracted from the clean speech signal and fed into the Extreme Learning Machine (ELM). The entire work is decomposed into three key phases and they are noise suppression, feature extraction and classification. The major contributions of this paper are listed below.

This work can effectively deal with noisy speech signals and thus the quality of the system is improved.

As the features are extracted from the cleansed speech signal, the recognition accuracy of the work is enhanced.

The remainder of this paper is organized as follows. The review of literature is presented in section 2. The proposed approach is presented in section 3. Section 4 evaluates the performance of the proposed approach. Finally, the concluding remarks are drawn in section 5.

2.0 REVIEW OF LITERATURE

This section intends to review the related literature with respect to noise suppression, feature extraction and classification.

2.1 Noise suppression

A stereo based block-wise linear compensation approach is proposed in [9], which dealt with babble, white and office noise. The work presented in [10] identifies speaker by utilizing spectral features. The spectral features are extracted by latent prosody analysis. In [11], a method to deal with stationary noise is presented, by utilizing short time Fourier transform and Ephraim-Malah estimation. This work reduces the spectral coefficients and thus the noise is suppressed. The work proposed in [12] tends to suppress noise by the transformation of the noise affected signal to the wavelet domain and thereby the local maxima coefficients are conserved. Level dependent threshold is introduced in [13], in order to eliminate coloured noise. In [15], the bionic wavelet transformation is improvised by clubbing the wavelet denoisin approach, in order to construct wavelet thresholding scheme. The exploitation of wavelet filters through multistage convolution withthe help of reverse biorthogonal wavelets in both high and low pass frequency bands.

However, it is not certain that the noise present in the training sample have to exist in the test speech signal. Though the existing works assume that both phases suffer from the same kind of noise and this reduces the recognition accuracy of the system. The proposed work aims to suppress the noise irrespective of the noise type and thus can serve its purpose effectively.

2.2 Feature extraction

Mel Frequency Cepstral Coefficients (MFCC) and Perceptual Linear Prediction (PLP) are the predominantly used features for the process of speech recognition [15,16]. The major drawback of MFCC is that the background noise can bring in the issue of disparity between the training and testing phases. The main factors for this issue are given below. The spectrum being assessed by MFCC is susceptible to noise and channel distortion [17]. Besides this, the acoustic model of MFCC is not found to be accurate for speaker recognition. Realizing the above mentioned points, the proposed work strives to utilize a feature which does not suffer from the background noise.All the stated drawbacks are overcome by Mean Hilbert Envelope Coefficients (MHEC). The performance of MHEC is proven to be the best under noisy environment and noise mismatch between training and testing phase.

Power Normalized Cepstral Coefficients (PNCC) is recently proposed and is proven to be better than Zero Crossing Peak Amplitude (ZCPA), Relative Spectral Transform – Perceptual Linear Prediction (RASTA-PLP), Perceptual Minimum Variance Distortionless Response (PMVDR) and so on [18-20]. PNCC proves its efficiency with different kinds of noise in both training and testing phase. However, it is a universal fact that hybridization of different algorithms improves the recognition accuracy even more. Taking the aforementioned statement into account, this work combines the two different techniques namely MHEC and PNCC, in order to combat against noise mismatch and to achieve higher recognition accuracy rate.

2.3 Classification

Classification plays a vital role in speaker recognition. Mostly employed classifier for speaker recognition system is Support Vector Machine (SVM) [21,22]. Some of the other classifiers are k-Nearest Neighbour (k-NN) and decision trees [23,24]. Besides this, certain generative models such as Gaussian Mixture Model (GMM) and Hidden Markov Model (HMM) are also employed. This work proposes to exploit ELM as the classifier, owing to its effectiveness, simplicity and speed.

3. Proposed approach

The objective of this section is to describe the functionality of the proposed speaker recognition system. The proposed work is decomposed into three building blocks, which are noise suppression, feature extraction and classification. The task of noise suppression is achieved by bionic wavelet transform. The process of feature extraction aims to excerpt MHEC and PNCC from the cleansed speech signal. This step is followed by the classification, which achieves the task of speaker recognition. The forthcoming subsections present the description of all the phases and the overall schematic diagram is presented in figure 1.

3.1 Noise suppression

A speech signal carries both essential and unnecessary details of information. This unnecessary detail of a signal can be denoted as noise. Thus, it is mandatory to suppress noise, which in turn improves the quality of the signal. However, noise suppression is a separate area of research and this work shows a small concern towards it. The proposed work employs bionic wavelet transform for noise suppression.

3.1.1 Bionic Wavelet Transform (BWT)

Basically, BWT follows the principle of Morlet wavelet, which is exclusively designed with respect to the human voiced signal [14]. The wavelet transform contains the coefficients of the signal $x(t)$, with regard to $h_{\alpha,\tau}(t)$ and all these are the elements of $h(t)$. This $h(t)$ is the mother wavelet and it is represented by

$$h_{\alpha,\tau}(t) = \frac{1}{\sqrt{|\alpha|}} h\left(\frac{t-\tau}{\alpha}\right) \quad (1)$$

The wavelet transform coefficients are obtained by the product of $x(t)$ and the basis functions,

$$W_x(\tau,\alpha) \leq x(t);\ h_{\alpha,\tau}(t) \geq \frac{1}{\sqrt{|\alpha|}} \int x(t) h^{*}\left(\frac{t-\tau}{\alpha}\right) dt \quad (2)$$

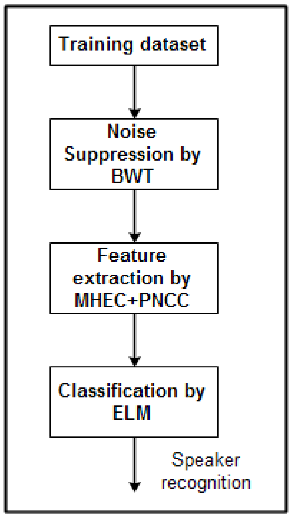

Fig 1: Overall schematic diagram of the proposed system

The basic idea of BWT is to substitute the constant quality factor with the changing adaptive quality factor [25]. This kind of substitution can be made by altering the mother function of the wavelet transform. The fluctuating $h(t)$ is denoted as

$$h(t) = \hat{h}(t) \exp\,(j2\pi f_0 t) \quad (3)$$

In the above equation, f_0 is the centre frequency of $h(t)$ and $\hat{h}(t)$ is the envelope function. T-value is employed in the BWT mother function and is represented in the below given equation.

$$h_T(t) = \frac{1}{T}\,\hat{h}\left(\frac{t}{T}\right) \exp\,(j2\pi f_0 t) \quad (4)$$

The BWT is represented by the following equations.

$$bwt_x(\tau, \alpha) = \frac{1}{\sqrt{|\alpha|}} \int x(t)\, h_T^{*}\left(\frac{t-\tau}{\alpha}\right) dt \quad (5)$$

This can be written as

$$bwt_x(\tau, \alpha) = \frac{1}{T\sqrt{|\alpha|}} \int x(t)\, \hat{h}^{*}\left(\frac{t-\tau}{\alpha T}\right) \times \exp\left(-j2\pi f_0 t \left(\frac{t-\tau}{\alpha}\right)\right) dt \quad (6)$$

From the above equations, it can be observed that the BWT mother function's amplitude and time spread depend on the value of T. The value of T can be generated on the basis of the idea of Yao and Zhang and is defined by

$$T(\tau + \Delta\tau) = \left(1 - \widetilde{G1}\, \frac{bwt_s}{bwt_s + |bwt_x(\tau,\alpha)|}\right)^{-1} \times \left(1 + \widetilde{G2} \left|\frac{\partial\, bwt_x(\tau,\alpha)}{\partial t}\right|\right)^{-1} \quad (7)$$

In the above equation, $\widetilde{G1}$, $\widetilde{G2}$ and bwt_s are constants, $bwt_x(\tau, \alpha)$ is the BWT coefficient at the time τ and scale α. Thus, it could be noted that the T function brings in adaptability to the BWT. The derivation of T function can be referred from [26,27]. The BWT coefficients are based on the WT coefficients and this can be represented by

$$bwt_x(\tau, \alpha; T) = k \times WT_x(\tau, \alpha) \quad (8)$$

The value of k depends on the value of T.

As already stated that the BWT is based on Morlet function, it is used as the mother function. The real morlet function is denoted by

$$h(t) = e^{-\left(\frac{t}{T_0}\right)^2} \quad (9)$$

The value of k is determined by eqn.10 as given in [28].

$$\frac{\int_{-\infty}^{+\infty} e^{-t^2} dt}{\sqrt{\left(\frac{T}{T_0}\right)^2 + 1}} \approx \frac{1.7725}{\sqrt{\left(\frac{T}{T_0}\right)^2 + 1}} \quad (10)$$

The noise suppression can be achieved by two levels of thresholds namely hard and soft thresholding techniques [29,30]. The degree of adaptability is based on the value of T function and so, T function is incorporated in determining the threshold. Besides this as per eqn.8, it is advisable to take least values for BWT coefficients, when the scales of decomposition are high. Thus, the T-function can be employed for lower scales of decomposition. The threshold can be calculated by

$$th = \frac{\sigma}{\sum_i \alpha_i T_{fs(i)}} \sqrt{2 \log_2 N} \quad (11)$$

The value of α_i is chosen by the trial and error method and it is more optimal to have decreasing function. After performing the operation of thresholding, the BWT coefficients are divided by the k factor and then inverse transform is performed. This rebuilds the signal to arrive at the cleansed signal. The next step is to extract features from the cleansed signal and is described below.

3.2 Feature extraction

Feature extraction is the most important phase of a speaker recognition system. This phase shows a great impact over the final classification stage. The better the choice of feature extractor, the greatest is the recognition accuracy rate. Thus, this work shows keen attention towards the
choice of feature extractors. After exhaustive study, we come to a conclusion that it would be optimal to choose feature extractors that can cope up with the noise mismatch during the training and testing phase. Understanding the benefits of hybridization, this work proposes to combine two feature extractors such as MHEC and PNCC and explained in the following subsections.

3.2.1 MHEC

The MHEC is recently proposed in [31], which effectively manages the noise mismatch in the training and testing stages. The cleansed signal $s(t)$ is passed for feature extraction, which is then divided into 24 bands with the help of a Gammatone filterbank [32]. The centre frequencies of the filterbank are placed on the rectangular bandwidth scale from 300 to 3400 Hz. The process involved in MHEC are presented in the below given equations.

$$s_a(t,j) = s(t,j) + i\hat{s}(t,j) \quad (12)$$

In the above equation, $\hat{s}(t,j)$ is the Hilbert transform of $s(t,j)$ and i is the imaginary part. The temporal envelope or Hilbert envelope $e_s(t,j)$can be c (13)computed by

$$e_s(t,j) = s^2(t,j) + \hat{s}^2(t,j)(13)$$

The duplicate high frequency components are suppressed by smoothening the temporal envelope with a low pass filter. The cut off frequency is set as 20 Hz.

$$e_{sn}(t,j) = (1-\alpha)e_s(t,j) + \alpha\, e_{sn}(t-1,j) \quad (14)$$

Where α is the smoothening factor and is inversely proportional to the cut-off frequency.

The resultant smoothened temporal envelope is then blocked into frames with duration25 ms and the skip rate is fixed as 10 ms. Each frame is then applied with a hamming window, so as to reduce the edge discontinuities. The temporal envelope amplitude in frame p is computed by

$$s(p,j) = \frac{1}{N}\sum_{t=0}^{N-1} w(t)\, e_{sn}(t,j) \quad (15)$$

$w(t)$ represents the hamming window and N is the size of the frame.

Natural log is applied to compress the spectral parameters. Besides this, Discrete Cosine Transform is applied to convert the spectrum to cepstrum and to decorrelate the feature dimensions. The outcome of this process is 36 dimensional feature vectors.

3.2.2 PNCC

The attractive features of PNCC are listed below. The introduction of its power-law nonlinearity, which is the substitute for log nonlinearity in MFCC paves way for robustness. The building blocks of PNCC are pre-processing, temporal integration, asymmetric noise suppression, temporal masking, spectral weight smoothening and mean power normalization.

Initially, a pre-emphasis filter which is of the kind $H(z) = 1 - 0.97z^{-1}$ is applied. This step is followed by the application of Short-Time Fourier Transform (STFT) by hamming windows. The factor $\hat{Q}[m,l]$ is only utilized for noise assessment and compensation. This can also be employed to alter the information with respect to the short time power estimates. This step is followed by the specifically designed asymmetric filter for noise suppression. Temporal masking is achieved by moving peak for every frequency channel and the power is suppressed, if at all it falls below the mask. The masked signals can serve its purpose in a better way, when it is exposed to reverberant environment. The outcome of the channels are then smoothened in spectral weight smoothening. Mean power normalization is incorporated into the system, so as to reduce the impact of amplitude scaling.

The noise mismatch is effectively handled by both MHEC and PNCC. Thus, efficient feature vectors are obtained. The so formed feature vectors are then fed into the classifier, in order to identify the speaker.

3.3 Classification by ELM

ELM is one of the effective classifiers with faster learning capability. The weights between the input and the hidden layer are allocated randomly. This can be achieved by a generalized inverse operation of the hidden layer output matrix.

Consider a training dataset (a_i, z_i) and $i = 1,2,..N$ and $a_i \in F^R$; $z_i \in F^q$. In this case, the ELM training is done by the following steps.

1. Allocate values to the lower layer weight matrix in a random fashion of the range [-1,1], such that $W \in F^{R.v}$, and v is the count of hidden units.

2.The hidden layer outcome is computed for each training sample a_i, such that $hl_i = \sigma(W^T\ a_i)$ in which σ is the sigmoid function.

3. The weight of output layer O is computed by

$O = (HH^T)^{-1}HT^T$ where $H = [h1, h2, \dots hn]$ and $T = [t1, t2, \dots tn]$

The random weight assignment is done irrespective of the training dataset and thus the new data can also be well-generalized. In this work, the ELM is trained with different voice samples, so as to classify between different speakers. The ELM's output for the test speech signal presents a k dimensional vector, which presents the top ranked ELM score from the trained dataset. The sample with maximum score is recognized as the speaker.

Thus, all the stages involved in the proposed work are described. The next section intends to evaluate the performance of the proposed approach.

4.0PERFORMANCE EVALUATION

The performance of the proposed work is analysed by exploiting KING Dataset [33]. The attributes of the dataset are given below.

Table 1: Details of king Dataset

Details	King dataset

Speaker count	51
Session count	10
Speech kind	Photograph description
Mic	Dual
Channel	Dual
Rate of sampling	8 kHz
Digital quantization	16 bits

While performing the experimental analysis, certain noises such as Babble noise, train, airport and car noise are included in the speech signal at different scales such as -5, 0, 5, 10 and 15 dB. All these noises are downloaded from the Noisex-92 database [32].

The signal quality is measured in terms of Signal-to-Noise Ratio (SNR) and Mean Square Error (MSE).

$$SNR = \frac{f^2}{n^2} \quad (16)$$

$$MSE = \frac{1}{N}\sum\left(f - \hat{f}\right)^2 \quad (17)$$

In the above equations, f is the clean signal and $\hat{f}$ is noised eliminated signal and n is the noise signal.

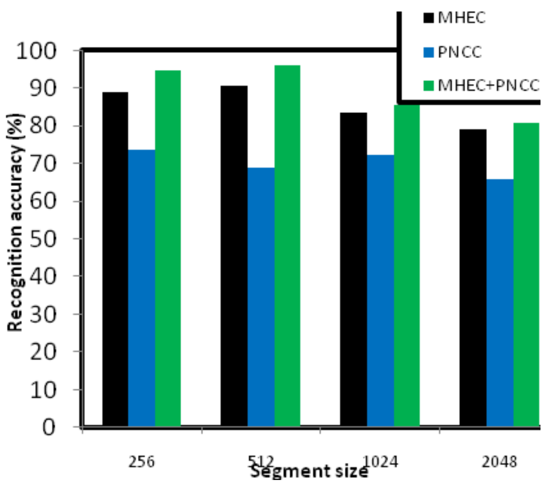

Fig 2: Recognition accuracy with k-means classification

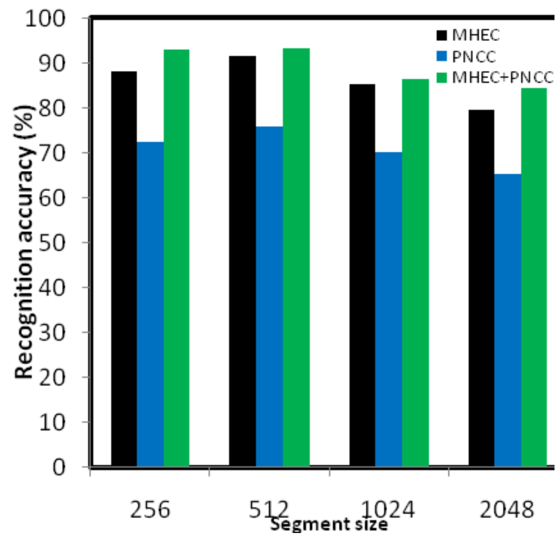

Fig 3: Recognition accuracy with SVM classification

The proposed approach is compared with two standard noise suppression methods such as spectral subtraction (SS) and Iterative Wiener Filtering (IWF) [32,34]. Besides this, we prove the effectiveness of the hybridization by carrying out the feature extractors individually. The overlap percentage is fixed as 60% and the results are obtained. Additionally, the comparison is made with respect to the classifier also. The proposed work is compared with k-means and SVM. Figure2,3 and 4 depicts the recognition accuracy rate ofthe system with respect to the k-means, SVM and ELM classification respectively.

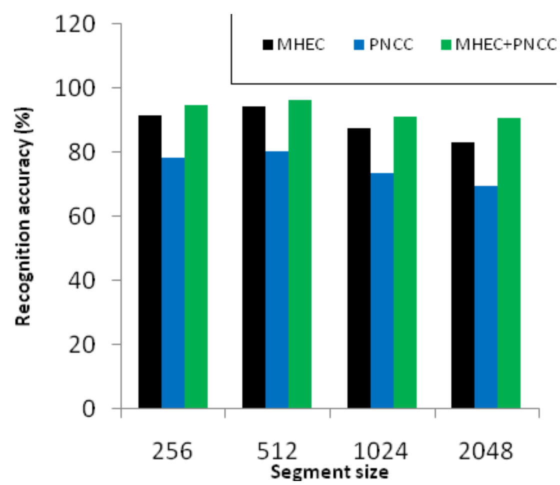

Fig 4: Recognition accuracy with ELM classification

On analysing the above graphs, it could be noted that the performance of the hybrid approach shows consistently good results. Besides this, the recognition accuracy rate of ELM is greater which when compared to k-means and SVM. The next part strives to compare the experimental resultswith respect to SNR and MSE and the graphical results are presented below.

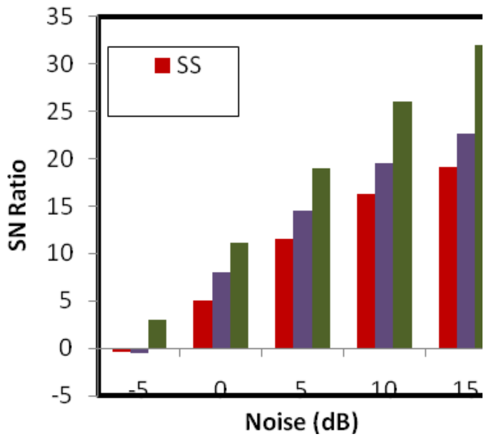

Fig 5: Signal to Noise Ratio

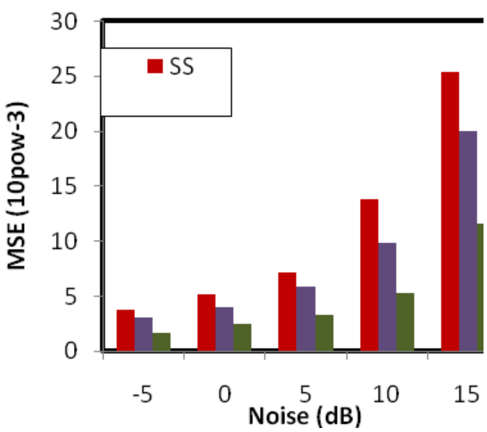

Fig 6: Mean Square Error

From the obtained experimental results, the performance of the proposed work is found to be satisfactory in terms of SNR, MSE and recognition accuracy rates. We found that the system performs well with the segment size of 256 and 512. Thus, the proposed work serves its purpose with greater accuracy rates.

5.0 CONCLUSION

This paper strives to present a novel speaker recognition system, which can effectively manage noise mismatch during training and testing phase. Initially, the noise is suppressed with the bionic wavelet transform with adaptive filtering technique. The obtained cleansed signal is then passed through the step of feature extraction. In this step, the features such as MHEC and PNCC are clubbed together, in order to inherit the merits of both the techniques. Finally, ELM is employed as the classifier to recognize between several speakers. On experimental analysis, the proposed work shows greater recognition accuracy rate and least MSE.

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Probe on Syntax Analyzer

Farhanaaz[1], Sanju. V[2] and **M. Vinayaka Murthy[3]**

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Abstract – Syntax analysis is the second phase of the compiler. Checks the syntactical structure of the programming language due to the limitation of the regular expression. Grammar is used to describe the syntax or rules of the source language. Parser is a tool used to check the syntactical structure and parsers are grammar specific. Though the significant work has be done on the parallel compilation process but still the parsing area is difficult to implement parallel on multi-core machines.

Index Terms – Parallel Compilation, parallel Syntax Analysis, Context Free Grammar, Top Down Parsing and Bottom up Parsing.

NOMENCLATURE

NOC - Network On Chip, CFG − Context Free Grammars, CFL − Context Free Language

1.0 INTRODUCTION

Grammar defines the syntactic structure of a programming language. Each grammar defines a unique programming language. Change in the grammar will result in programming language. Roles and responsibilities of the syntax analyzer [1][2][3][4] are :

(i) To take token as a input from lexical analyzer
(ii) To check if tokens could be generated from the specified grammar of the programming language.
(iii) To report syntactical errors in the program if any.
(iv) To construct parse tree.

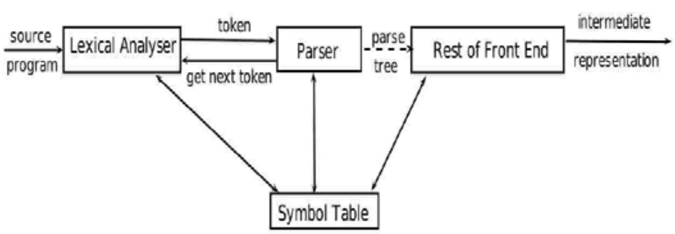

Figure 1: Position of a Parser in Compiler

[1] Assistant Professor, School of Computer Science and Applications, REVA University, Bangalore, India, Email: farhanaaz3@gmail.com
[2]Associate Professor & Head, Department of CSE, Muthoot Institute of Technology and Science, Ernakulam, India, Email: sanjuv21@gmail.com
[3]Professor & Assistant Director, Research and Innovation, REVA University, Bangalore, India, Email: vinayakamurthy@revainstitution.org

2.0 GRAMMAR

A grammar is the powerful tool for describing the language. Grammars are language generators. Noam Chomsky gave the mathematical model for the grammars in 1956.

Though it can't describe natural languages but it is very useful to describe computer languages. There are different types of grammar.

(i) Type 0: unrestricted grammar include all formal grammars. The languages generated by this grammar is known as recursively enumerable languages.
(ii) Type 1: Context Sensitive Grammar generate Context Sensitive Languages which is recognized by Non Deterministic Turing Machine.
(iii) Type 2: Context Free Grammar generate Context Free Languages which are recognized by Non Deterministic Pushdown Automaton.
(iv) Type 3: Regular Grammar generate regular languages and it is recognized by Finite State Automaton.
Out of these Context Free Grammars are used in syntax analyzer to define a structure of a language.

Class	Grammar	Languages	Automaton / Machine
Type 0	Unrestricted	Recursively Enumerable	Turing Machine
Type 1	Context Sensitive	Context Sensitive	Linear Bound
Type 2	Context Free	Context Free	Push Down Automata
Type 3	Regular	Regular	Finite

Table 1: Types of Languages and their Acceptable Machines

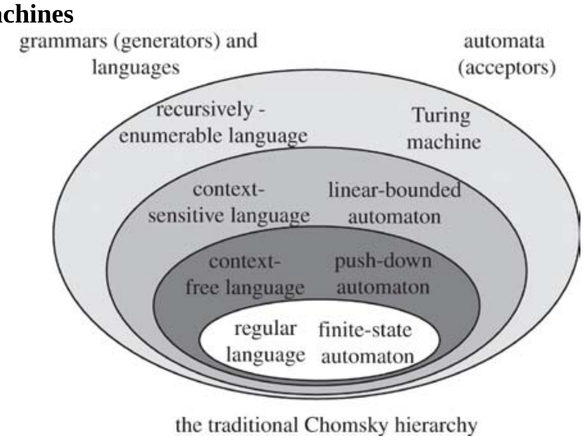

Figure 2: Chomsky Hierarchy

2.1 Context Free Grammar

Formally Context Free Grammar is define by G = (N, T, P, S)
N : Finite set of Non Terminals; generally represented by Upper case alphabets.
T : A finite set of Terminals represented by Lower case alphabets.
S : Starting Non Terminal symbol of the grammar. $S \in N$
P : Set of rules or productions in Context Free Grammar, each of the form $D \rightarrow \alpha$ where $D \in N$ and $\alpha \in (N \cup T)*$. First production always indicates the start symbol of the grammar. Below is an example of CFG and the derivation of the string from the given productions.
$S \rightarrow 0S0/1S1/0/1/\varepsilon$.
Let us derive a string 1001001. Let us start with the appropriate production
$S \rightarrow 1S1 \rightarrow^{S \rightarrow 0S0} 10S01 \rightarrow^{S \rightarrow 0S0} 100S001 \rightarrow^{S \rightarrow 1} 1001001$.
Superscript production indicates the application of that production in next step. CFG generate CFL. The grammar generated by G is represented by L(G). L(G) = { w | $w \in T*$, and $S \Rightarrow * w$} .

2.2 Parse Tree

Parse Tree is the pictorial representation of a derivation. A parse tree is an ordered tree in which nodes are labeled with the left side of the production and in which children of the nodes represents its corresponding right side. Except root and leaf nodes of the tree others are all non terminals therefore productions are applied to replace the non terminals with the RHS of the production and the leaf nodes are all terminals. Formally Parse tree is defined as, If grammar G is the CFG then G = (N, T, P, S). If G is the derivation tree if and only if
(i) The root is the Start symbol.
(ii) Internal nodes are Non terminals from N.
(iii) Leaf nodes are Terminals from T.
(iv) If leaf node is # then it has no siblings.
Yield of the parse tree is the list of labels of all the leaf nodes from left to right. If α is the yield of derivation tree for grammar G , then $S \Rightarrow * \alpha$

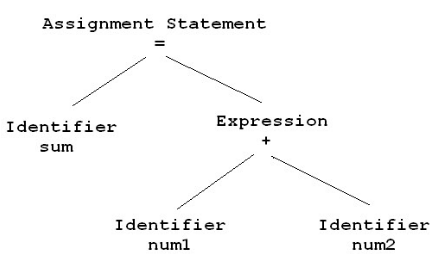

Figure 3: Parse Tree

2.3 Left and Right Linear Grammar

If all the productions in the CFG are in the form A → Bw / w then it is known as left linear Grammar. If the productions are of the type A → wB / w then it is a right linear grammar. A and B are variable and $w \in T*$.

Left most and Right most derivation

To restrict the number of choices while deriving a string we opt for left most and right most derivation. A derivation is said to be left most iff the left most non terminal is replaced by the appropriate production till the string is formed. Likewise in the right most derivation the right most non-terminal is replaced with the appropriate production. Left most and right most derivations can be derived for the string aabbaa and the grammar is S→ aDS / a and A → SbD / SS / ba. Left most Derivation : S→ aDS → aSbDS → aabDS → aabbaS → aabbaa. Right most Derivation : S→ aSD → aDa → aSbDa → aSbbaa → aabbaa.

2.4 Issues in writing a Context Free Grammar for programming language

1) Elimination of Ambiguous grammar: A grammar is ambiguous, if for at least one string in the language, grammar produces more than one parse tree. Derive a^3 using grammar S→ aS / Sa / a Sometimes ambiguity can be eliminated by rewriting the grammar. For the simplicity purpose we can restrict the format of the Context Free grammar without reducing the language generation power. Let L be a non-empty CFL, then CFL can be generated by a CFG G with the following properties :
(i) Elimination of Useless Symbols : variables or terminals that do not appear in any derivation of a terminal string from start symbol.
(ii)Elimination of ε productions : If production of the form D→ε for some variable D.
(iii) Elimination of unit productions : If productions of the form D→E for variables D and E.

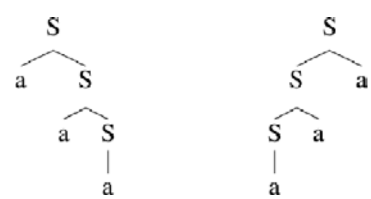

Figure 4: Different Parse Tree for a^3

2) Elimination of left Recursion: Recursive non terminals are very useful which allows grammar to describe infinite number of input but left recursive grammars couldn't be handled by top down parsing techniques. A grammar contains a production of the form $A \rightarrow A\alpha$, where A is non terminal; then this production can be replaced by a non left recursive production of the form $A \rightarrow \beta B$ and $B \rightarrow \alpha$, $B \rightarrow \varepsilon$, without changing the strings derivable from A. This grammar is of the type Left recursive. This procedures remove left recursion from A to B generating same language as A. This procedure does not eliminate left recursion involving derivations of more than 2 steps. Above procedure can be extended to n number of variables on left hand side of the production.
3) Elimination of Left Factoring: Left factoring is the powerful tool in generating grammar which is accepted by predictive parsers or top down parser. Suppose we have the production of the form $A \rightarrow \alpha\beta_1 / \alpha\beta_2$, then on seeing the

input α in the production A it is not clear which production is of right choice, this can be rewritten in the form $A \rightarrow \alpha B$ and $B \rightarrow \beta_1 / \beta_2$.

3.0 PUSHDOWN AUTOMATON

Finite Automaton cannot store / remember anything. Therefore Finite Automaton can be extended by adding auxiliary storage to accept CFG. Push down automaton is a finite automaton with control of both an input tape and a stack to store.

Formally, PDA is defined as a Finite Automaton P = (Q, Σ, Γ, S, F, δ)

Q : Non empty finite set of states

Σ : Non empty finite set of input symbols

Γ : Stack alphabet

S : Initial State, $S \in Q$

F : Non empty finite set of Final State/(s) and $F \subseteq Q$

δ : Transition Function which maps to $(Q \times \Sigma^* \times \Gamma^*) \rightarrow (Q \times \Gamma^*)$.

Moves in Push Down Automaton

(i) $\delta(q, a, z) \rightarrow (p, y)$: If PDA is in the state q, with z as the top of the stack and with a on the input tape then PDA replaces z by y on top of the stack and enters state p.

(ii) $\delta(q, a, \varepsilon) \rightarrow (p, a)$: Push a on to the stack.

(iii) $\delta(q, a, z) \rightarrow (p, \varepsilon)$: Pop element from stack.

(iv) $\delta(p, a, z) \rightarrow (p, z)$: PDA does nothing.

Push Down Automata can recognize languages for which there exist Context Free Grammar.

4.0 PARSER

Parser is the program for parsing. Parsing is the technique which it produces an output as a parse tree for the input string w. An error message will be indicated if w is not a valid for the given grammar, otherwise parse tree is generated. Parsing is classified based on the rules implemented to arrive at the solution. Following are the types

4.1 Top Down Parsing

In a top down approach, a parser starts constructing a parse tree from the top node called root node and it completes the parse tree in pre order fashion for the given input string. Top down parsing holds the technique form leftmost derivation for an input string. The types of top down parsing is depicted in Figure6.

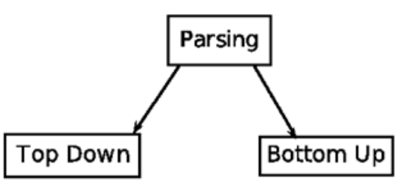

Figure 5: Types of parsing.

1) Recursive-descent parsing: This is one of simplest form of top down approach. The program consist of a set of procedures, one for each non terminal of the grammar. Execution begin with the process for the start symbol, which stops and announces hit if its procedure scans the entire input string. Following is the procedure for the non terminal A in the grammar.

```
void A(){
Choose an A production, A→ X1 ,X2 . . . Xk ;
for( i = 1 to k){
if(Xi is a non terminal)
call procedure Xi ();
else if(Xi equals the current input symbol a)
advance the input to the next symbol; else
Error occurred; }
}
```

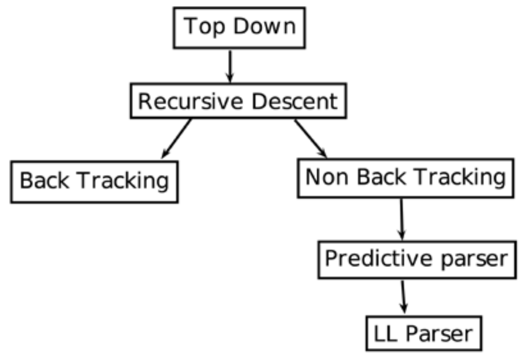

Figure 6: Types of Top down Parsers

General recursive-descent may need backtracking technique for repeated scans over input to arrive at the correct input. To allow backtracking the above code, the code needs to be modified in such a way that, it not only checks for current non terminal but also for all non terminals available in grammar to find the correct productions which matches with the input string, if it does not match then it raises an error.

Back Tracking Technique: Every string generated by applying productions on trail and error method based on the input string matched. If the prediction of the production is successful then parsing continues, otherwise in case of mismatch then at this stage previous prediction has to be rejected and pointer has to be set to the previous position and next production is predicted. This is known as Backtracking. Backtracking is one of the major drawback of top down parser. Predictive parser is the efficient non backtracking form of top down parser, where lookahead symbol unambiguously determines the procedure for each non terminal and hence no backtracking occurs.

Following example demonstrates the Backtracking technique. Consider the following grammar

$S \rightarrow hQf$

$Q \rightarrow al/a$

Using above grammar evaluate string w=haf

Step 1: S is the start symbol, therefore grammar starts from the symbol S and has only one production.

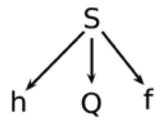

Figure 7

Step 2: Now input pointer is set to Q because h is a terminal. At this stage Q tends to 2 production that is Q → al/a. Parser predicts the production and Q is expanded.

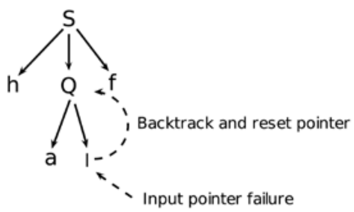

Figure 8: Backtracking

Step 3: Correct alternative is predicted and yield of the parse tree is w = haf.

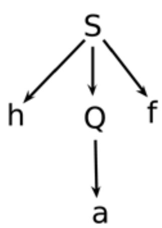

Figure 9: Alternative production

2) Predictive Parser: Recursive Descent parser which needs no back tracking is called predictive parser. Predictive parser technique can exactly decide that which production to be used based on the next input symbol. Predictive parser program maintains a stack which hold only non terminals and uses two dimensional table created from grammar.
Parser acts on the basis of two symbols that a symbol on top of stack and look ahead pointing to input buffer. Based on the various possibilities
1. If Stack top \$ = lookahead symbol \$ then parser halts. Successful Parsing Condition.
2. If Stack top is a terminal. Stack top t = lookahead symbol t then parser pops t and advances lookahead pointer, otherwise an error is raised.
3. If Stack top is a non terminal then parser predicts the entry. Non terminal is popped from stack and the right side of the production is pushed on to the stack from left to right. If appropriate production is not present then parser raises an error.
Predictive Parsers can be constructed for the class of grammar called LL(1). LL(1) grammar covers most of the programming constructs.
FIRST and FOLLOW Computation : FIRST and FOLLOW are the two necessary preliminary functions which is used in LL grammar. These functions allows us to select which productions to apply, based on the next input symbol.

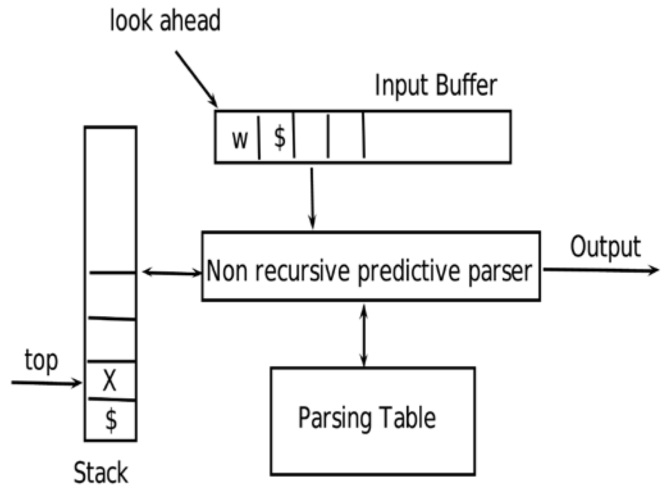

Figure 10: Model of Non recursive Parser

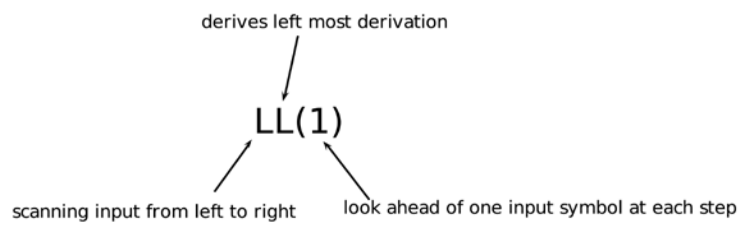

Figure 11: Meaning of LL(1)

FIRST : is function which gives the set of terminals that begins the strings derived from the production rule. Formally FIRST(α) = { t / (t is the terminal and $\alpha \Rightarrow^* t\beta$) or (t→ ε and $\alpha \Rightarrow^* \varepsilon$)}
FIRST Computation : To define FIRST(α). Let us define for a single symbol D
1. If D is a terminal : FIRST(D) = D.
2. If D is ε : FIRST(D) = ε.
3. If D is non terminal : In this case we must look at all grammar productions with D on left. If production is of the form $D \rightarrow Y_1Y_2\ Y_3 \bullet \bullet \bullet Y_n$, where Y_i is single terminal or non terminal. a is in FIRST(D), if for some i, a is in FIRST(Y_i)
and ε is in all of FIRST(Y_1) • • • FIRST(Y_i) that is Y_1 • • •
$Y_{i-1} \Rightarrow^* \varepsilon$. If ε is in FIRST($Y_j$) where j = 1,2,...,n then add ε to FIRST(D). Everything in FIRST(Y_1) is surely in FIRST(D). if Y_1 does not derive ε then we add nothing more to FIRST(D) but if $Y_1 \Rightarrow^* \varepsilon$ then FIRST(D) = FIRST(Y_1) -{ε} ∪ FIRST(Y_2) and same method is applied to subsequent non terminals.
FOLLOW : is function which gives the set of terminals that can appear immediately to the right side of the given symbol. It is defined for the single non terminal. Formally : FOLLOW(A) = {t / (t is the terminal and $S \Rightarrow^+ \alpha At\beta$) or (t

is EOF and S⇒* α A)}. FOLLOW Computation : To define FALLOW(A). A is a single non terminal. FOLLOW(A) = EOF, if A is the start non terminal. For each production X→ α A β put FIRST(β) - {ε} in FOLLOW(A). if ε is in FIRST(β)then put FOLLOW(X) into FOLLOW(A). For each production X→ α A, put FOLLOW(X) into FOLLOW(A).
Construction of Parsing Table.
Parsing Table is constructed based on the function call select. Select function can be defined by First and Follow function. If production is of the form A→X then Select(A→X) = FIRST(FIRST(X) x FOLLOW(A)).
A Context Free grammar whose parsing table has no multiple entries is said to be LL(1). If LL(1) has same entries then the grammar is ambiguous and/or left recursive and/or not left factored. LL(1) Property : If non terminal appears on the left side of more than one production and select for those productions are disjoint. If this property hold good for a given grammar then grammar is LL(1). Following is the example to check id following grammar is LL(1).
E → PX
X → +PX / ε
P → RK
K → *RK / ε
R → (E) / id
FIRST and FOLLOW Computations : Compute FIRST and FOLLOW Functions for all the non terminals. Both are computed based on the given definitions above.
FIRST(E) = FIRST(PX) because : E →PX
= FIRST(P)
= FIRST(RK)
= FIRST(R)
= {(, id }
Likewise for the other productions FIRST function is computed
FIRST(X) = {+, ε}
FIRST(P) = FIRST(R) = {(, id }
FIRST(K) = {*, ε}
FIRST(R) = {(, id }
Now Compute FOLLOW function
FOLLOW(E) = { $,) }
FOLLOW(X) = FOLLOW(E) = { $,) }
FOLLOW(P) = FIRST(X) ∪ FOLLOW(E) = { +, ε } - {ε} ∪ { $,)} = { +,), $ }
FOLLOW(K) = FOLLOW(P) = { +,), $ }
FOLLOW(R) = FIRST(K) ∪ FOLLOW(P) ∪ FOLLOW(K) =
{*, ε} -{ε} ∪ { +,), $ } ∪ { +,), $ } = { *, +,), $}
Now Construct Parsing Table
To Construct parsing table which is also called as M table, we need Compute Select Function which guides us to fill the table. Select function is defined by FIRST and FOLLOW functions. For any production say X→W. Select function for the given production is defined as
SELECT(X→W) = FIRST(FIRST(W) x FOLLOW(X))
Let us compute Select functions for all the productions
SELECT(E→PX) = FIRST(FIRST(PX) x FOLLOW(E)) = FIRST({(, id} x {$,)}) = FIRST{((, $), ((,)), (id, $), (id,)} = {(, id}.
Similarly for the other productions SELECT function is computed.
SELECT(X→+PX) = FIRST(FIRST(+PX) x FOLLOW(X)) = FIRST({+} x { $,) }) = { + }
SELECT(X→ε) = FIRST(FIRST(ε) x FOLLOW(X)) = FIRST({ε} x { $,)}) = {$}
SELECT(P→RK) = FIRST(FIRST(RK) x FOLLOW(P) = {(, id}
SELECT(K→*RK) = { * }
SELECT(X→ ε) = { +,), $}
SELECT(R→(E)) = { (}
SELECT(R→id) = {id}
Following is the Parsing Table
The above table is filled on the following basis. For the production E →PX, SELECT(E) = { (, id } in this case the corresponding entry for M[E, (] = PX and M[E, id] = PX. Likewise other entries are made in the table based on Select function.

Non-terminals	(	id	+	*	)	$
E	PX	PX				
X			+PX		ε	ε
P	RK	RK				
K			ε	*RK	ε	ε
R	(E)	id				

Table 2: Predictive Parsing Table

LL(1) property : A grammar is an LL(1) iff the parsing table has no entries that are multiply defined. If a non terminal appears on the left side has more than one production then SELECT for those productions are disjoint, this is LL(1) property. For the same grammar above, non terminals which has more than one production are X, B and F.
For production X→+PX, X→ε SELECT function for X,
SELECT(X→+PX) ∩ SELECT(X→ε) = ϕ
For production K→*RK andK→ε SELECT(K→*RK) ∩ SELECT(K→ ε) = ϕ
For production R→(E) and R→id SELECT(R→(E)) ∩ SELECT(R→ id) = ϕ
Therefore X, K and R have LL(1) property. The given Grammar is LL(1).

4.2 Bottom Up Parsing
In bottom up approach, Parser starts constructing parse tree from the leaf node and works towards root node. Simplest form of Bottom up parsing is Shift Reduce Parser.
1) Shift Reduce: Shift reduce parser reduces the given input string into the start symbol. This parser uses 2 unique steps,

namely shift and reduce step. Data Structure used in this parser are Stack, input buffer, data structures to store and access the left and right of the production. Shift reduce parser performs various actions.

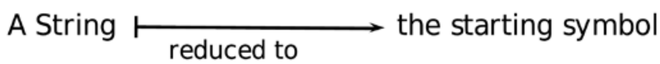

Figure 12: Shift reduce parser

(1)Shift action: Parser shifts the input symbol from the input tape on top stack one symbol at a time.
(2) Reduce action: It reduces top of the stack using appropriate production. The reduction is performed by popping the right side of the rule from the stack and pushing the left side of the production.
(3) Accept action: Parser announces the successful parse if the stack contains the start symbol and input tape is empty then input is accepted.
(4) Error action: If parser is not able to shift or reduce or accept, it announces syntax error has occurred.
Initial Configuration: $ is push on to stack to mark end of the stack and $ is concatenated at the end of the input string to indicate the end of string.
Limitations of Shift Reduce Parser :
(1) Shift Reduce Conflict : If Context Free Grammar has 2 productions of the form $A \rightarrow \beta$ and $\beta \rightarrow \beta p \gamma$. If B is on top of the stack and next token is p then parser is not able to decide whether it has to shift or reduce. This is known ans shift reduce conflict.
(2) Reduce Conflict: If Context Free Grammar has 2 productions of the form $A \rightarrow \alpha$ and $B \rightarrow \alpha$. If α is on top of the stack then in this case parser is not able to decide which production to apply to perform reduce action. This situation is Reduce Conflict.
2) LR Parser: LR parser is a non recursive, shift reduce, bottom up parser. LR grammars are a subset of CFG for which LR parsers can be constructed.

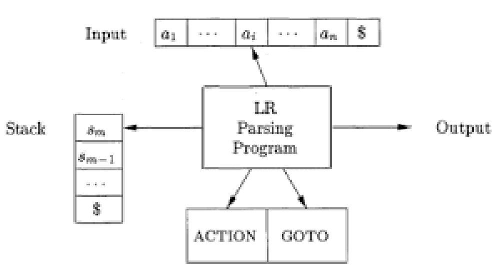

Figure 13: Model of LR Parser

LR parser require input, output, a stack, driver program and a parsing table. Parsing table consist action and goto procedure. Driver program remains same for all parsers, only parsing table changes according to the grammar. The parsing program reads characters from an input buffer one at a time. A program uses a stack to store a string of the form $S_0 X_1 S_1 X_2 \cdots S_m X_m$ where S_m is on top of the stack and X_i is grammar symbol and S_i is a State. If S_m is top of the stack and a_i is the current input symbol then driver program perform action[S_m, a_i] procedure which can one of the following actions
(1) Shift S, where S is the state action[S_m, a_i] = shift s, parser executes the shift move entering the configuration. A configuration of an LR parser is a pair whose first component is a stack content and second component is a the input.
(2) Reduce by the grammar production $A \rightarrow \beta$ action[S_m, a_i]=reduce $\rightarrow \beta$
(3) accept action[S_m,a_i] = accept, paring successfully completed.
(4) error action[S_m, a_i] = Error, parser discovers an error and calls for error recovery routine.

5.0 PARALLEL SYNTAX ANALYZER

Research was started on parallel compilation with the advent of microprocessors early in 1970 where Lincoln[12] first proposed the idea of parallel object code. Later Zosel[5] recognized the parallel loops. Mickunas and Shell[6] recognized the area in a compilation method where parallelization can be achieved and also proposed parallel lexical analysis, where lexical analysis can be broken into 2 sections called scanning and screening. They also proposed a parallel parsing method based on LR parsing. The 2 major requirement of the parallel processing is determining the ends of the reducible phrase and performing reduction parser was also extended, called piecewise LR (PLR). Many researchers attempted many other techniques to achieved parallelism during compilation process. Parallel syntax analyzer was implemented on different files[7]. This was achieved by selecting the file and scheduling to the specific processor for syntax analysis using processor affinity[8]. To estimate the speed up[9] in parallel processing 3 different modules were written.
(1) A Simulator, which emulates the behavior of the processor.
(2) A Generator, which keeps track of time as simulator works.
(3) An Estimator, computes the approximate numbers of basic parsing operation.
To compare the performance with that of parallel compilation in multi-core with respect to single core, Jacqus, Hickey and Joel[10] Computed upper Bounds for Speed up gained synchronous, multi purpose, bottom up, no back tracking parsing generated by bottom up parsing along with few assumptions made by them. Issues in implementing Parallel parsing on multi core machines was also identified[11]. Issues are division of code and Synchronization, Processor issues, Threading, Task distribution and Context Switching.

6.0 CONCLUSION

Though the work has been done but still the significant research has to carried out in this field to parallel syntax analysis. Various attempts has been made to parallelize parsing but still issues exists. Major work has to be done in identifying the area to be parallelized, splitting the code and synchronizing it. Future work is to develop syntax analyzer for NoC architecture.

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Optimization of KNN with Firefly Algorithm

Alka Lamba[1] and **Dharmender Kumar**[2]

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Abstract – Data mining has turned out to be a milestone in information industry. The need of data mining tools can be evidenced in almost every field. Classification is one of the data mining techniques which are used for knowledge discovery. Out of the various alternatives to evolve a classification model, KNN is a very popular and apprehensible one. Although, KNN incorporates a number of limitations in it but these can be bumped-off by making some alterations to the standard KNN algorithm. Numerous variants of KNN have been proposed by many researchers in previously done studies and they have also outperformed the standard KNN. In present study, a modified version of KNN algorithm has been proposed which commingles firefly algorithm with standard KNN. The performance of this modified algorithm is examined with respect to the standard KNN and it is found that the proposed algorithm works well in case of large data sets.

Index Terms – classification; data mining; firefly; KNN; self-adaptive

1.0 INTRODUCTION

Today the data is increasing by leaps and bounds. The availability of abundant amount of data has increased the necessity of data mining tools. These tools help in exploring data in such a way that it results in obtaining some crucial information. These results of data mining can be utilized to make important decisions in various fields such as marketing, financial data analysis, medical science, intrusion detection, retail industry etc[1],[2],[3]. Data mining offers different data mining techniques which are used for mining knowledge from data i.e. clustering, classification, association rule, prediction, outlier detection. An introduction to these data mining techniques and their applications are given in [4]. Classification is one of the data mining techniques which are used frequently. In this data mining technique, a classification model is built which is called classifier. The model is capable of classifying a data tuple. Classification constructs this classifier using a class-labeled data set. This is the reason why classification is said to be an example of supervised learning. The process of classification starts with partitioning of the data set into two sets: training set and test set.

[1]Master of Technology (CSE), Guru Jambheshwar University of Science and Technology, Hisar, Haryana, India.
[2]Associate Professor (CSE), Guru Jambheshwar University of Science and Technology, Hisar, Haryana, India.
[1]alkalamba07@gmail.com and
[2]dharmindia24@gmail.com

After this, two steps are followed. The construction of a classifier using a pre-classified training data set is the first step and the assessment of the constructed classifier using test set is the second step. The second step testifies that how good is the built classifier in predicting class labels for unknown tuples. There are several classification algorithms to carry out the process of classification for example k-Nearest Neighbor (KNN), Bayesian Classifier, Decision Tree Induction, Support Vector Machine (SVM), Classification using Back-Propagation, Rule-Based Classification etc. [5]. A comparative study of these classification algorithms is projected in [6]. Out of these, KNN is the simplest and most comprehensible. KNN employs the nearest neighbor technique where the classification of an unseen tuple is done using similar data tuples to it. K-Nearest Neighbor classifies a data tuple on the basis of class-labels of the k nearest data tuples to it in the data set. The k is assumed to be a positive integer and passed as input to the KNN algorithm. KNN algorithm is a lazy learner with non-parametric nature [7]. Unlike parametric methods the non-parametric methods does not make any presumption about the shape of the classification model. The reason of categorizing KNN as a lazy learner and rest of the classification algorithm as eager learners is that KNN does not construct any classifier as done normally by other classification algorithms on getting training data tuples. All the calculations are done only at the time of classification of an unseen tuple. On account of this working principle of KNN, another name of it is instance-based learner.

Besides simplicity, there are many plus points of KNN. It is scalable. No prior knowledge is required by it regarding the data set. It gives quite good results when compared to the results given by other classification algorithms. Insensitivity towards the noisy data adds another element to its list of merits. Despite of a number of positives, many negatives are associated with KNN. k is the only parameter that KNN takes as input. But it a very challenging task to determine the appropriate value of k. It is so because taking the small value of k would increase variance of the obtained model and large value of k would increase bias of the resulting model [8]. And it is well- known that we have to get a trade-off between variance and bias to construct a good model. Another issue with KNN is that it requires a lot of memory to store the training tuples due to its lazy nature. This can be managed in case of small data sets but with large data sets it becomes very nasty. The large computational cost is another demerit of KNN. The cause of this increased computational cost is the manner in which KNN classifies a data tuple. In KNN, for classification of a novel tuple the k nearest tuples to it are needed to be determined. And this is accomplished by computing the distance of the novel tuple with all the training tuples. It raises the computational cost of the algorithm.

Owing to these infirmities of KNN, many researchers have proposed reformed versions of KNN. Different variants of KNN are discussed in [9]. These variants tried to alleviate the shortcomings of KNN algorithm. The results of KNN algorithm are influenced by numerous key factors such as the value of k, distance metric which is used for computing similarity between any two tuples, the weights of attributes in a data set. By manipulating these key factors the performance of KNN can be ameliorated. [10] chooses the value of k by consulting local neighborhood of the data tuple which is to be classified. In the primitive KNN all the attributes of data set give equal contribution in classification of unprecedented tuple. But not all the attributes are significant. This can contaminate the results of KNN. [11] proposes an algorithm which is called weight adjusted KNN. In this proposed algorithm the enumeration of weights for each of the feature in the data set is done. [11] exerted the proposed weight adjusted KNN for text categorization. [12] uncovered the aspect that the classes in the data set are not evenly dispersed. There can be more number of tuples embraced in one class than other. By dint of which the outcome can be biased towards the class which encompasses larger number of tuples. [12] suggested a solution for this by using different values of k for distinct classes according to the number of tuples contained by them. Small value of k is used for the class that has fewer tuples and large value of k is used for the class that has large number of tuples. To lessen the computational cost incurred in KNN, an amended variant of KNN called adaptive KNN is proffered in [13]. It uses a non-fixed value of k instead of a fixed value along with some early-break heuristics. [14] discusses various extensions for KNN which are density- based KNN classifier, variable KNN classifier, weighted KNN classifier, class-based KNN classifier and discernibility KNN. All of them use different methodology to uproot the flaws in standard KNN. Another variant of KNN is presented in [15] which performs classification of a data tuple using shared neighbors. To gauge similarity between any two tuples, it uses BM25 similarity measure. To confine the number of neighbors that can vote for classification of a novel tuple a threshold is set. An amalgam of clustering algorithm K-Means with KNN is proposed in [16]. This amalgamation tried to reduce the computational cost of KNN. To enhance the accuracy attained from standard KNN, distance metric plays a very vital role. Standard KNN generally employs Euclidean distance metric. A new distance metric is introduced in [17] which is called Mahalanobis distance metric. The advantage of using this distance metric in place of Euclidean is that the correlation between data tuples is also reckoned by it. One more distance metric, informativeness is introduced in [18]. The algorithm proposed in [18] takes two parameters as inputs which are k and I. Firstly, k nearest neighbors to the unseen data tuple is determined. After this, informativeness for each nearest neighbor is evaluated. Out of these, only I most informative data tuples are considered to vote in classifying any tuple. [19] proposed an algorithm which is combination of a number of KNN classifiers. Each classifier is trained on different part of data set. To find the class of a test tuple, the class of the test tuple is enumerated using all classifiers. The majority class will be the class of test tuple.

Soft computing which offers information processing when united with data mining in a creative way, then this formation can be used efficaciously for knowledge discovery in large databases. [20], [21], [22], [23] elaborates that how soft computing helps in carrying out a better data analysis. Many data mining tasks can be expressed as optimization problems such as feature selection, clustering, classification etc. And soft computing can be used to find approximate solutions for these optimization problems. The principal components of soft computing include Fuzzy Logic, Rough Sets, Neural Networks and Evolutionary Computing. Further evolutionary computing comprises of two kinds of algorithm: Evolutionary algorithms and Meta-heuristic algorithms [24]. Evolutionary algorithms include genetic algorithm and differential algorithm whereas meta-heuristic algorithms embrace cuckoo search, particle swarm optimization (PSO), firefly algorithm, ant colony optimization (ACO), artificial bee colony (ABC), Bayesian network etc. [25] discusses about all these nature-inspired meta-heuristic algorithms. There are various studies that have been carried out earlier which demonstrate that how well these meta-heuristic algorithms perform in case of classification. [26], [27], [28] have deployed Ant Colony Optimization algorithm for classification. [29],[30] have used Particle Swarm Optimization algorithm for classification. Artificial Bee Colony algorithm is used for image classification in [31]. In [32] a hybrid fuzzy firefly algorithm is used to derive classification rules. [33] proposed a fuzzy classification system. Some of these principal components of soft computing are infused with KNN in earlier done studies. [34] proposed a fuzzy version of KNN algorithm. In contrast to KNN, which gives crisp membership of the tuples, it gives fuzzy membership. A fuzzy-rough nearest algorithm is proposed in [35] which combines rough set theory and fuzzy set theory. The proposed algorithm employs Rough set theory to compute the lower and upper approximations of classes using nearest neighbors [36]. Test tuple is classified based on its membership in these approximations. Being a powerful optimization tool [37], genetic algorithm has also been implanted with KNN. A hybrid version of KNN with genetic algorithm is proposed in [38]. Instead of using any distance metric, it utilizes genetic algorithm for determining the k nearest data tuples. [39] has used genetic algorithm with KNN differently. It exercised genetic algorithm for extracting worthy features from a data set. [40] indulges ACO algorithm with KNN to pick out good features from a data set. PSO algorithm has been integrated with KNN divergently. In [41], PSO is used to find representatives of distinct classes in the data set. The representatives will now be the new training data tuples and KNN will be implemented on these new tuples for classification of a novel tuple. [42] made use of PSO for figuring out weights for features in the data set. ABC algorithm has also been used similar to that of PSO with KNN. [43] practiced ABC for extracting good features from a data set and [44] implemented ABC to find representatives of distinct classes as done in [40].

[43] put to use the combination of ABC and KNN for diagnosing coronary heart disease.
After studying the erstwhile studies and weaknesses of KNN, it can be concluded that standard KNN algorithm can be refined further in order to grab good accuracy. Taking inspiration from these, this study proffers another variant of KNN which would conglomerate firefly algorithm and KNN. Firefly algorithm is inspired from the flashing behavior of fireflies. [45], [46], [47], [48] scrutinizes the performance of firefly algorithm. Many variants of firefly algorithm can be seen in former studies. A randomization term is needed in firefly algorithm which comprises of two parameters α and ε. α is called randomization parameter and it decides the next place to search for solution in the search space or explicitly it defines the step size for a firefly. ε is a vector of random numbers which is originated from a probability distribution method. There are numerous methods to draw this vector such as uniform distribution, Gaussian distribution, levy flights distribution etc. [49] proposed a variant of firefly algorithm called Levy Flights Firefly algorithm. The algorithm draws the random number vector using Levy Flights distribution. The firefly algorithm makes an assumption that if all the fireflies have same brightness then they will move randomly. [50] suggested that rather than moving randomly they should move towards the global best. Also it employed Gaussian distribution for drawing ε. Another variant of firefly was proposed in [51] which is called self-adaptive step firefly algorithm. The self-adaptive step firefly algorithm computes step size for each firefly according to its fitness values in previous generations. In addition to flashing light of fireflies the algorithm propounded in [52] considers some other affecting parameters also i.e. gene exchange of firefly, its pheromone and the dispersion of the pheromone due to wind. In this research paper, self-adaptive step firefly algorithm is opted to infuse into standard KNN algorithm.
In the subsequent sections of present research paper we will learn about KNN, firefly algorithm and then the proposed modified KNN algorithm. The performances of the KNN and the propounded algorithm will be monitored on six data sets. The data sets are taken from UCI Repository and Keel.

2.0 K-NEAREST NEIGHBOR ALGORITHM

KNN algorithm is a very popular classification technique. It can be put into practice very easily. The prerequisites for the KNN algorithm are: a class-labeled data set and the input parameter k. The value of input parameter k would resolve that how many nearest neighbors are to be taken into account for classification of any tuple. The procedure of classifying any tuple using KNN is straightforward. Initially, the data set is bifurcated. The two subsets are called training set and test set. The part of both is same as they have in classification. Later on k nearest data tuples to unseen tuple from the training set are determined. The class which has majority in these unearthed k data tuples is assigned to the unseen tuple, which is to be classified. Test set will compute the accuracy of the KNN algorithm. Pseudo code for the standard KNN algorithm is given below:

ALGORITHM I

Input Parameters: Data set, k
Output: Classified test tuples
Step 1: Store all the training tuples.
Step 2: for each test tuple

A. Compute distance of it with all the training tuples using (1).
B. Find the k nearest training tuples to the test tuple.
C. The class which is most common in the k nearest training tuples to the test tuple is assigned to the test tuple.

End for

Each tuple in data set can be viewed as a data point in the n-dimensional space, where n is the number of attributes describing the data set. The distance between the data points is computed generally using Euclidean distance. Euclidean distance between two data tuples x and y is given below:

$$\sqrt{\sum_{1\le i\le n}(x_i - y_i)^2} \quad (1)$$

n = number of attributes in data set
x_i and y_i are values of attribute i in data tuples x and y respectively. Manhattan distance and Minkowski distance are some other distance metrics which can also be used.
The simplest case of k-nearest neighbor algorithm is when k is taken to be 1. This case is called nearest neighbor rule, where the class assigned to the unseen tuple is the class of most nearest tuple to it. Another property of KNN is that it can be employed not only for predicting a categorical attribute but also for predicting a continuous valued attribute. The later one is called regression. In regression, the value of class attribute of an unseen tuple will be the average of the class attribute values of the k nearest tuples to the unseen tuple.

3.0 FIREFLY ALGORITHM

Firefly algorithm is meta-heuristic in nature and is used to find an approximate solution for an optimization problem. Flashing behavior of the fireflies is inspiration of the firefly algorithm. There are three assumptions made in the firefly algorithm:

- Any firefly can be attracted towards any other firefly.
- The attractiveness is relative to brightness of the firefly. Brighter firefly would attract all other fireflies having less brightness than the brighter firefly.
- When all fireflies have same brightness then they will move randomly.

The attractiveness of a firefly is calculated using following function:

$$\beta(r) = \beta_0 . e^{-\gamma . r^2} \quad (2)$$

where β_0 is the attractiveness of the firefly when r = 0 and γ is light absorption coefficient. The firefly's movement totally depends on its attractiveness. Firefly i would move towards firefly j if and only the attractiveness of the firefly j is greater than that of firefly i. In that case, the movement is shown by following formula:

$$x_{ik}= x_{ik}+ \beta_{0.}e^{-\gamma.r_{ij}^2}.(x_{ik} - y_{jk}) + \alpha.S_k.(rand_{ik} - 0.5) \quad (3)$$

$x_{ik}$and y_{jk} are values of attribute k. k takes values from 1,2………n, where n is the dimension of the data set. $rand_{ik}$ is a random number between 0 and 1. α is called randomization parameter which will decide how much to move and takes value between 0 & 1. $S_k$is scaling parameter which is calculated for each attribute. $S_k$is calculated as

$$S_k= | u_k - l_k | \quad (4)$$

$u_k$and$l_k$are the upper bound and lower bound of the attribute k respectively. $r_{ij}$is the distance between the fireflies i and j which calculated from:

$$r_{ij} = \sqrt{\sum_{1 \le i \le n}(x_i - y_i)^2} \quad (5)$$

The value of attractiveness in optimization problems is calculated using an objective function. The algorithm for standard firefly algorithm is given below:

ALGORITHM II

```
Input: Objective function f(x) and algorithm
        parameters α0, β0 and γ
Output: Minimized function value position x_min
Step 1: Initialize firefly population  p randomly.
Step 2: Initialize algorithm parameters α0,  β0 and γ.
Step 3: Calculate fitness value using the objective function
f(x) for each firefly.
Step 4: while ( t <maxgeneration )
            for i=1:p
              for j=1:i
                 if ( f(x_j) < f(x_i) )
                  move firefly i towards j using  (3)
                  calculate fitness value again of all
                  fireflies
                 end if
              end for
           end for
        end while
Step 5: Rank the fireflies to find the current best firefly.
```

4.0 KNN WITH FIREFLY ALGORITHM

As discussed before, there are many variants of firefly algorithm available in antecedent studies. Self-adaptive step algorithm is one of them. A comparison of the performances of self-adaptive step firefly algorithm and the standard firefly algorithm is demonstrated in [32]. The obtained results demonstrate that self-adaptive firefly algorithm is better than standard firefly algorithm in every aspect. The self-adaptive step firefly algorithm and standard firefly algorithm differs at randomization parameter α. In standard firefly algorithm the parameter α is either fixed all time or decreases exponentially. But in case of self-adaptive step firefly algorithm α is calculated for each firefly according to the fitness values that it has attained previously. The notion behind is that a firefly which is far from the global best solution should take larger steps and the firefly which is near to the global best solution should take smaller steps so that it can converge slowly to give best results.

$$h_i(t) = 1/\sqrt{(fit(t-1) - fit(t-2))^2 + 1} \quad (6)$$

$$\alpha(t+1)=1-1/\sqrt{(fit_{best}(t) - fit(t))^2 + (h_i(t))^2 + 1} \quad (7)$$

Here, fit(t-1) = fitness of the firefly in $(t-1)^{th}$ generation.
fit(t-2) = fitness of the firefly in $(t-2)^{th}$ generation.
fit_{best} = fitness value of the best firefly in $(t-1)^{th}$ generation.
fit(t) = fitness of the firefly in t^{th} generation.

This conglomerate of KNN and self-adaptive step firefly algorithm works as follows: The foremost task is to reckon the representative of each distinct class in the data set using self-adaptive step firefly algorithm. After accomplishing this, the process of classifying any tuple becomes very easy. These obtained representatives would now be acting as new training data tuples. And when a job of classifying any unseen tuple is assigned we have to just calculate its distance from these new training data tuples only. The unknown tuple is categorized in that class, the representative of which has the least distance with that unknown tuple. The pseudo code for the proposed algorithm is given below:

ALGORITHM III

```
Step 1: Normalize the data set.
Step 2: Find representatives of each class in data set using
          Self-adaptive step firefly algorithm and in order to
fulfill this, follow the subsequent steps.
        a)  Initialize algorithm parameters   ,     and γ
            and input objective function f(x).
        b)  Divide the training data set according to the
            class attribute.
        c)  for each  training data set grouped via class
            attribute.   Let n be the number of fireflies in
            set.
                while ( t <maxgeneration)
                  for i=1:n
                    for j=1:i
                      if ( f(x_j) < f(x_i) )
                      move firefly i towards firefly j
using equation (3). Calculate α
using formulas in (6) and (7).
                      end if
                    end for
                  end for
                end while
        d)  Find the current best firefly and choose it as
            representative of that class.
        e)  for each test tuple
              calculate the distance of the test tuple
from each of class representatives. Assign
              that class to the test tuple from whose
representative it has the least distance.
```

There are many advantages of this reoriented KNN algorithm. The first one is you don't need to pass the input parameter k anymore as we have to do in standard KNN. Ascertaining the appropriate value of the parameter k is itself a challenging task. The second benefit is, it would reduce the cost complexity of KNN algorithm. This optimized KNN would sustain for longer period because once you have computed the representatives of each distinct class then the task of classifying a tuple would take only a fraction of seconds. On the other hand in case of KNN algorithm the cost complexity was high because for classification of a tuple, you have to compute its distance with all the training tuples every time.

5.0 EXPERIMENT AND RESULTS

The performance of the proposed modified KNN and the standard KNN is tested on six data sets of different sizes. The data sets are picked from UCI repository and Keel. Their performances are summarized in form of tables TABLE I and TABLE II. The TABLE I depicts the algorithmic parameters taken for self-adaptive step firefly algorithm, the number of generations for which the firefly algorithm is run. The proposed algorithm and the standard KNN algorithm both are implemented in MATLAB software. Hold out method has been used to split up the data set into training and test sets. 30% of the data set is upheld as test data set and rest of the data set is used to train the model. Three parameters are used to compare performances of both the algorithms which are accuracy, time taken for classifying all the test tuples and kappa statistics. The performances of both the algorithms in aspects of accuracy and time are shown with help of graphs. Fig. 1 depicts that the proposed algorithm gives accuracy comparable to that of standard KNN in case of large data sets. From Fig. 2 it can be seen that the proposed algorithm takes much less time in classifying the test tuples when compared to standard KNN algorithm and the difference enlarges when the size of data set is large.

6.0 CONCLUSION

In present research paper, a modified KNN algorithm is proposed which has used self-adaptive step firefly algorithm to find representatives of distinct classes in data set. This study demonstrates that the proposed algorithm optimize the results by taking much less time in comparison to standard KNN. Due to which the cost of computation also got reduced. On scrutinizing the results obtained, it can be concluded that the proposed algorithm performs well in case of large data sets.

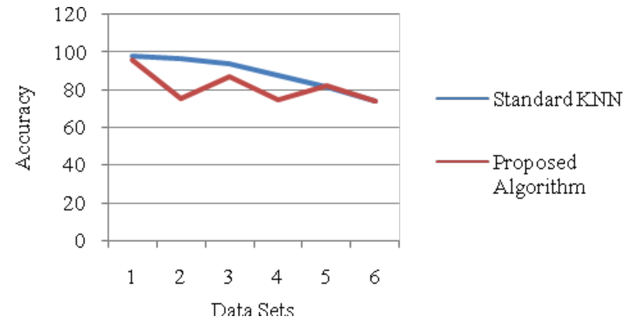

Figure 1: Graph depicting accuracies attained by both the algorithms

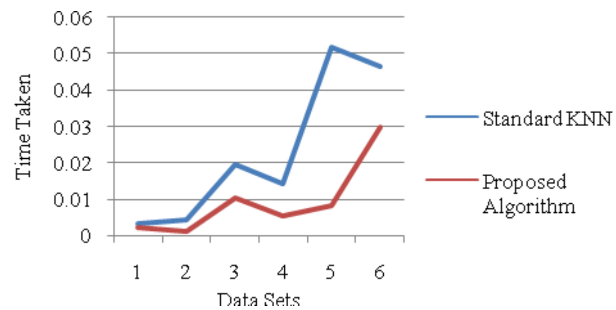

Figure 2: Graph depicting time taken by both the algorithms to classify test tuples

Table 1: Experiment Results for the Proposed Algorithm

Data sets	No. of instances in data set	Number of classes	Algorithmic parameters				Accuracy(%)	Time taken to classify test tuples(sec)	Time saved In comparison to Standard KNN	Kappa Statistics
			α_0	β_0	γ	Number of generations				
Iris	150	3	0.9	1	0.1	50	95.55	0.002110	39.57%	0.93
Wine	178	3	0.9	1	0.1	60	75.00	0.001160	74.15%	0.63
Wholesale	440	2	0.9	1	0.1	50	87.02	0.010322	47.34%	0.72
Data user modeling	404	4	0.9	1	0.1	70	74.79	0.005516	61.58%	0.66
Australian	690	2	0.9	1	0.1	50	82.04	0.008165	84.21%	0.63
pima	768	2	0.9	1	0.1	50	73.91	0.029670	35.87%	0.43

Table 2: Experiment Results For The Standard Algorithm

Data sets	No. of instances in data set	k	Number of classes	Accuracy(%)	Time taken to classify test tuples(sec)	Kappa Statistics
Iris	150	5	3	97.78	0.003492	0.97
Wine	178	5	3	96.15	0.004487	0.94
Wholesale	440	5	3	93.89	0.019602	0.86
Data user modeling	404	5	4	87.39	0.014356	0.83
Australian	690	5	2	81.55	0.051713	0.63
pima	768	5	2	73.91	0.046269	0.42

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Lexical, Ontological & Conceptual Framework of Semantic Search Engine (LOC-SSE)

Gagandeep Singh Narula[1], Usha Yadav[2], Neelam Duhan[3] and **Vishal Jain[4]**

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Abstract – The paper addresses the problems of traditional keyword based search engines that process query syntactically rather than semantically. In order to increase degree of relevance and higher precision to recall ratio, it describes proposed architecture of Semantic Search Engine (SSE) which incorporates Google search results as input and processes them with the help of Semantic Web (SW) technologies. Modules to accomplish various tasks like query processing, importing existing ontologies and extraction of knowledge have been introduced in proposed framework. At last, the PROMPT algorithm is being applied to compare query graph and document graph which leads to improved results that are presented to user..

Index Terms- ***Semantic Web (SW), Ontology, PROMPT, Protégé 3.4.8, Jena, Resource Description Framework (RDF) and Knowledge Retrieval***

1.0. INTRODUCTION

Traditional search engines are tools for retrieving information from massive sources on the web.The results are being produced by performing keyword based search most of the tme.The main drawback of search engines is lack of relevance. To illustrate the problem in a better way, consider a query "**Mobile phones with red cover**" submitted to a traditional search engine. It produces relevant as well as irrelevant results in relation with terms-mobile phones, red lotus, flower and cover. The search experience does not consider stopping words, auxiliary verbs that reflects the meaning of given statement. Likewise in above query, the term "with" has lost its significance due to which results are being produced in context of lotus and red flower. In order to reduce this ambiguity and perform intelligent search, concept of Semantic Web (SW) came into existence in 1996 as envisioned by Tim Berners Lee [1]. SW is defined as global mesh of information in machine interpretable format [2].It is practically not feasible to annotate the entire web content into semantic tags so that current search engines could behave like Semantic Search Engines (SSE).

[1]Research Scholar, M.Tech (CSE), CDAC Noida
gagan.narula87@gmail.com
[2]Assistant Professor, CDAC-Noida
usha.yadav.912@gmail.com
[3]Assistant Professor (CE), YMCA University of Science & Technology, Faridabad, India neelam.duhan@gmail.com
[4]Assistant Professor, Bharati Vidyapeeth's Institute of Computer Applications and Management (BVICAM), New Delhi
vishaljain83@ymail.com

So, there is need to develop search engine that analyses user query and produces meaningful results with higher precision and low recall.

The following paper is categorized into following sections. Section 2 describes objective and scope of research carried out in given paper. Section 3 presents brief survey of research conducted in context of evolution of SSE's and their methodologies. Section 4 provides bird's eye view of Semantic Web layered architecture and comparative analysis of studied literature survey. Section 5 describes proposed SSE framework along with its implementation. Section 6 validates higher precision to recall ratio in comparison to GOOGLE. Section 7 concludes the given paper followed by references.

2. 0. OBJECTIVE, SCOPE & FINAL OUTCOME

Objective

"To enhance GOOGLE [2] search results with the help of Semantic Web technologies".

Scope

A user would be able to learn about semantic web technologies, semantic web tools, ontology development for knowledge representation and storing that knowledge using some open source framework.

Final Outcome

The intended final outcome of work carried out is precise and relevant search results produced by enhancing GOOGLE search results with the employment of SW technologies.

3.0. RELATED WORK

Several studies that have been conducted with an aim to build SSE and ranking of results as follows:

Debajyoti et.al [3] proposed semantic search framework that produces relevant results by performing mapping between classes and instances with the help of RDF codes. Fatima et.al [4] adds query optimizer, user interface and processor in its framework but it too has some limitations. Zhang et.al [5] performed keyword based search by finding RDF files and compares keywords with its contents. Swati et.al [6] proposed information retrieval system in context of university domain but it does not evaluate GOOGLE search results. Kumar et.al [7] made use of mapper and query processor for representation and scanning of keywords respectively.

For comparative analysis of these works, refer to Table 1.

4.0. SW ARCHITECTURE

According to Kevin Kelly [8], it suffers from fax effect which means that development of semantic web is costly and its technologies have not been utilized fully. But, still most of researchers are trying their hands on this web technology to achieve machine- human interaction [8]

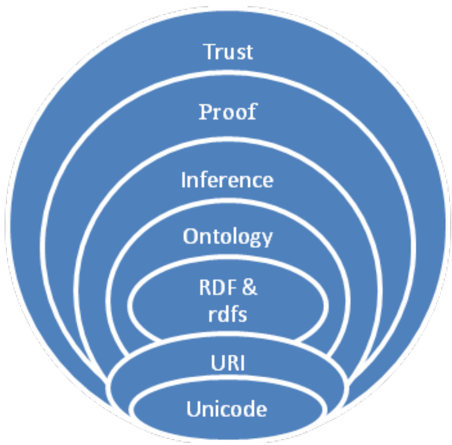

Figure1: SW architecture [9]

Table 1: Comparative Analysis

Research Work	Pros	Cons
Debajyoti et.al [3]	(a) Uses ontology to maintain semantic relationships among classes and instances rather than using NLP. (b) Values of property can be computed from RDF codes and displayed to user	(a) No ranking of results is being done.
Fatima et.al [4]	(a) Query optimizer scans keywords and matches them with words stored in ontology database.	(a) No updating of ontology database. (b) User interface is not connected to any semantic framework.
Zhang et.al [5]	(a) Combines Google search results with RDF and present them in hierarchical fashion. (b) OntoSearch acts as visualization tool and can be linked to other web ontology editor tools	(a) Synonym problem is not well addressed in this version of tool
Swati et.al [6]	(a) Uses WordNet API for generation of semantically similar words. (b) Matches terms used in user query with designed ontology to produce refined query.	(a) Does not evaluate Google results.
Kumar et.al [7]	(a) Uses Mapper to represent semantic results into textual format. (b) Query processor scans keywords and matches them with words stored in ontology database.	(a) No comparison and evaluation of IR performance. (b) Does not evaluate Google results

5.0. COMPONENTS OF PROPOSED SSE FRAMEWORK

The proposed framework as outlined in Fig 2. consists of three phases:

- Generation of user query graph with the help of SW technologies.
- Generation of document relation graph by analyzing GOOGLE search results.
- Comparison of source and target ontologies that leads to improved results

First phase

(a) GUI: - The interface on which search is performed is treated as main component of any search engine. In context of traditional search engines, queries are written by developers and results are matched with pre-defined keywords stored in databases. But in proposed work, ontology is used as backend in interface.

In given framework, input query is being passed through user interface as well as GOOGLE search engine. It is passed to search engine in order to enhance search results with the help of SW technologies.

(b) Designing /Importing existing ontology: - The proposed framework uses PROTÉGÉ 3.4 beta [10] for importing existing ontology related to given domain. *Protégé is an open-source tool for editing and managing Ontologies. It is the most widely used domain-independent, freely available, platform-independent technology for developing and managing ontologies.*

(c) Extracting knowledge from given ontology: - Apache JENA framework can be used to represent relationship between classes, properties and instances from given ontology. It will lead to formation of knowledge base. *JENA is a java framework for building semantic web applications that provides programmatic environment for RDF, RDFS, and OWL and consists of rule based inference engine [11].*

Second Phase

Same user query is being entered in GOOGLE search engine and results are retrieved. These results are in form of HTML (Text) documents. So, relationship among those text documents is extracted by converting them into RDF documents. It is done with the help of Text2RDF application.

Third Phase

This phase requires comparison of target ontology graph and source ontology graph. In both graphs, concepts are represented by nodes while relations are represented by edges. It is done with the help of PROMPT [12] algorithm. Features of PROMPT are as:

- Besides merging ontology, it identifies locations for integration of ontologies, type of operations to be performed and resolves conflicts.
- Interactive merging process i.e. several choices are being performed by user and PROMPT selects them automatically on basis of user preferences.

- Handle conflicts like name conflicts, dangling references, redundant classes and slot value restrictions.

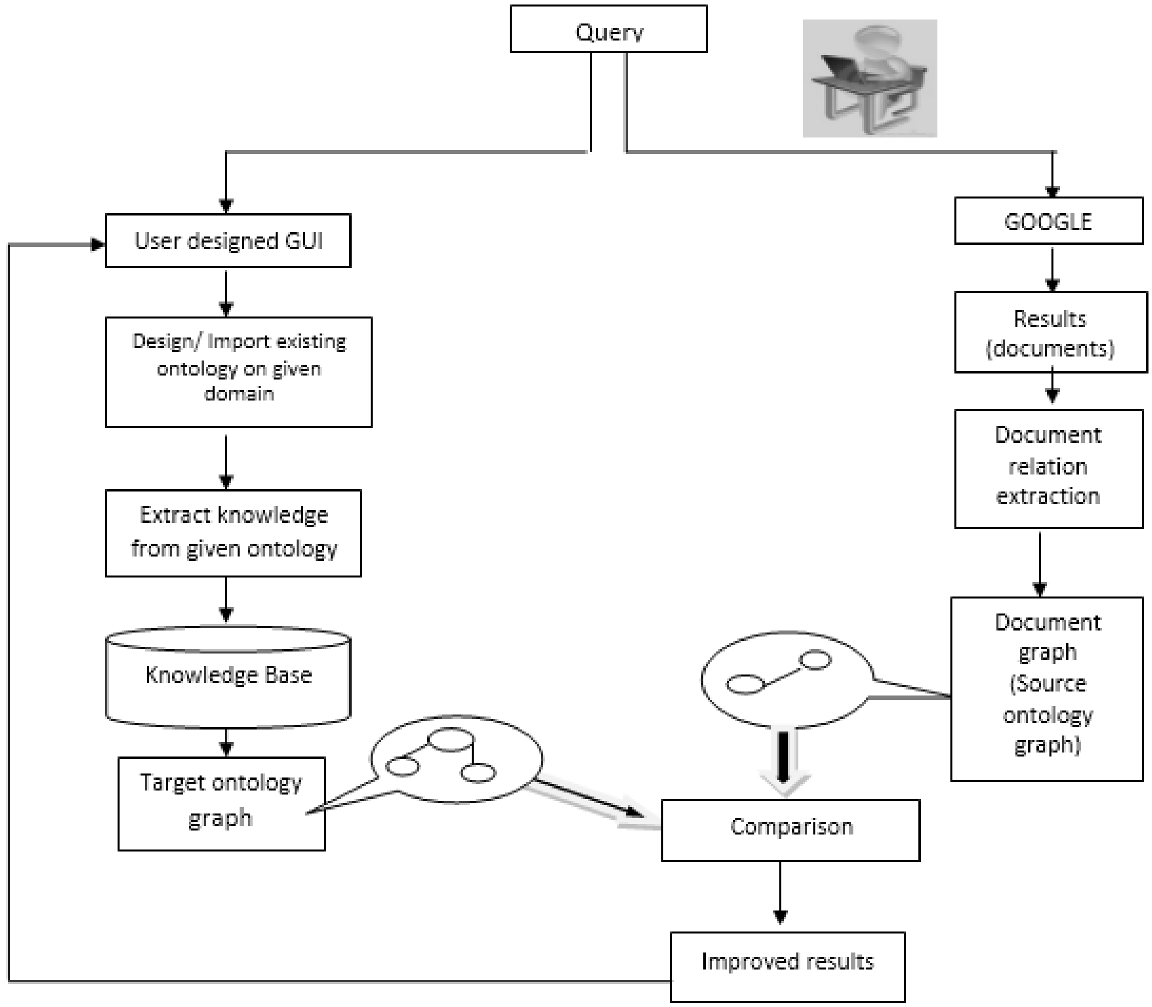

Figure 2: Proposed LOC-SSE framework

5.1. Pros of Proposed Approach

- The given framework evaluates GOOGLE search results in addition to user query.
- User interface is connected to semantic framework called as JENA in order to retrieve knowledgeable results from ontology.
- Relationships among classes, properties and instances are represented in form of user query graph.
- On other hand, document graph is being created from GOOGLE search results.
- Thus, above methodology adds *lexical, conceptual and ontological* flavor to proposed framework.

5.2. Implementation

Above approach is being implemented as shown in steps below:

Consider user query as "**List the faculties of CSE in IIIT Hyderabad**"

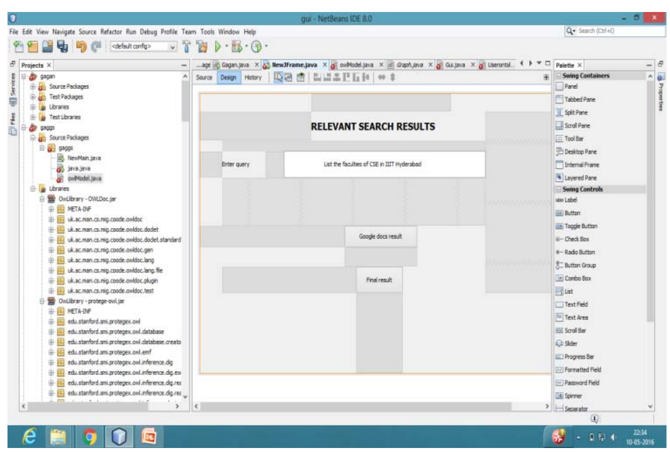

Figure 3: Home Screen

Step 1

(a) User designed GUI: This form is drawn in NetBeans IDE 8.0

(b) Showing data connectivity among Protégé, NetBeans IDE and Jena

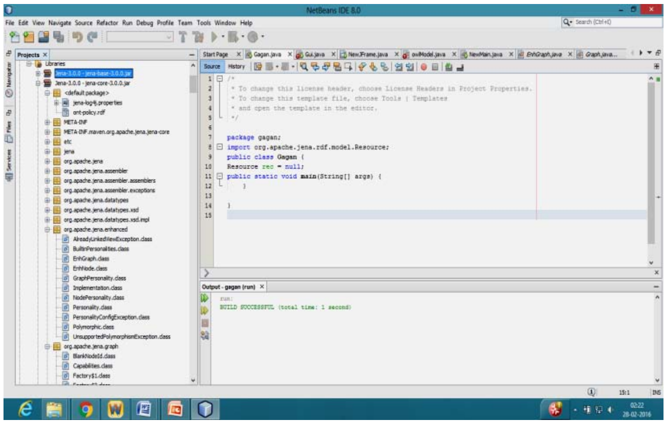

Figure 4: Importing libraries & its successful execution

Step 2: Designing of ontology on given domain (Educational_institute.owl)

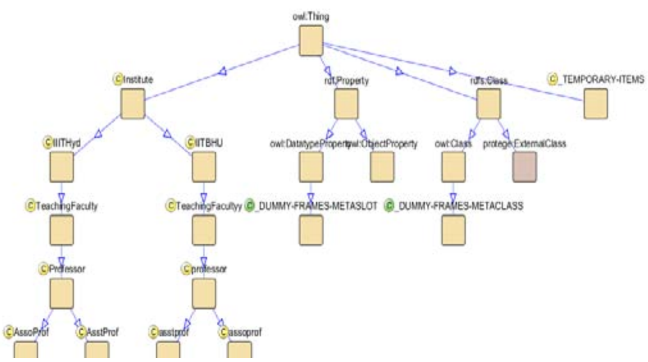

Figure 5: Educational_institute domain ontology

Step 3: Extracting knowledge from given ontology

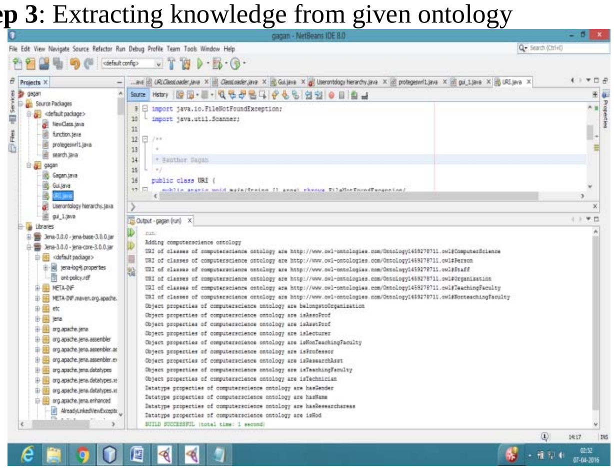

Figure 6: Displaying properties and URI's of Educational_institute ontology

From Fig 5, subsection of target ontology has been extracted further on basis of query "List the faculties of CSE in IIIT Hyderabad".

```
SELECT ?Institute
WHERE
{
<http://www.owl-ontologies.com/Ontology1460357725.owl#Institute> ?Institute
}
PREFIX abc:<C:\Program Files (x86)\Protege 3.4.8>
SELECT ?name
WHERE
{
?abc:name?IIITHyd
}
    }
```

Figure 7: Extracting target ontology portion using SPARQL query

Step 4: Creation of knowledge base involves:
Generation of rules using Semantic Web Rule Language (SWRL)
Four rules are being created that can lead to inferences related to given query.

(i) Rule1_//_Hod_is_AssoProf_whose_Name_is
Its expression in SWRL is
CSE:isAssoProf(?A, ?S) ∧ CSE:isHod(?H, ?A) → CSE:hasName(?H, ?S)

(ii) Rule2_//_AssoProf_is_senior_to_Lecturer_and_AsstProf
Its expression in SWRL is
CSE:isAsstProf(?A, ?S) ∧ CSE:isLecturer(?L, ?A) → CSE:isAssoProf(?L, ?S)

(iii) Rule3_//_AsstProf_for_TeachingFaculty
Its expression in SWRL is
CSE:isTeachingFaculty(?G, ?F) → CSE:isAsstProf(?F, ?G)

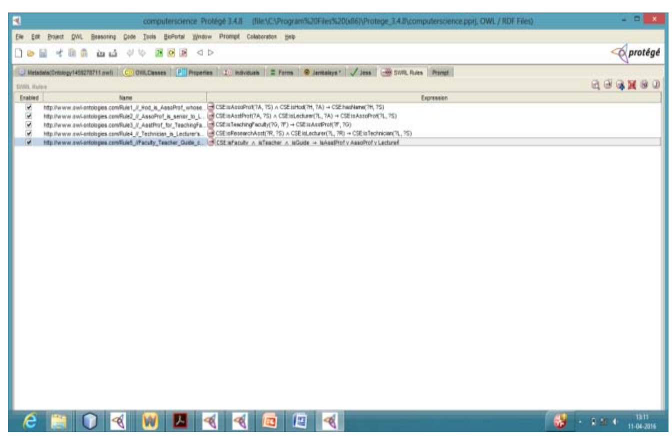

Figure 8: Rules generated using SWRL

Step 5: Target ontology graph

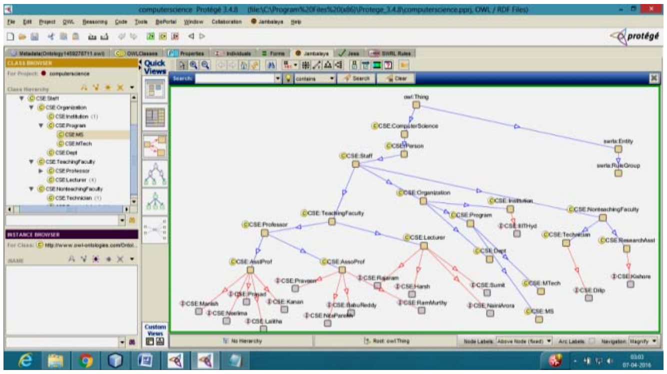

Figure 9: Target ontology graph

Step 6: Now, query is entered on Google and it produces links of other faculties of IIT BHU, IIT Hyd in addition to IIIT Hyd

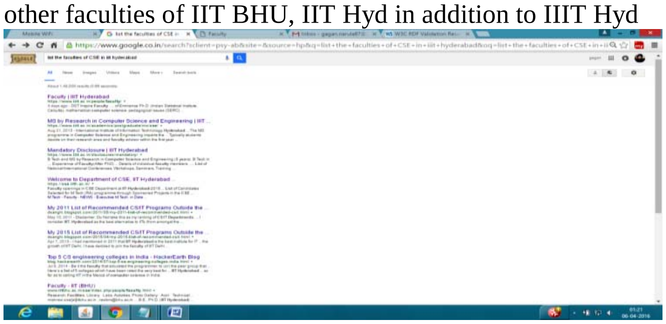

Figure 10: GOOGLE search results page

Step 7: Results (documents)
It involves conversion of HTML links to semantic web resources like RDF so that ontology can be created which can be said as “GoogleCSE.pprj” or “GoogleCSE.owl”

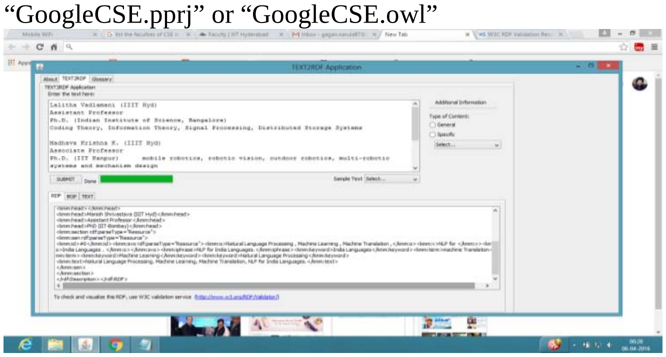

Figure 11: Conversion of HTML links of IIIT Hyd into RDF

Similarly, conversion of IIT Hyd and IIT BHU can be done into RDF.

Step 8: Document Relation Extraction
It involves designing of ontology from above RDF results.

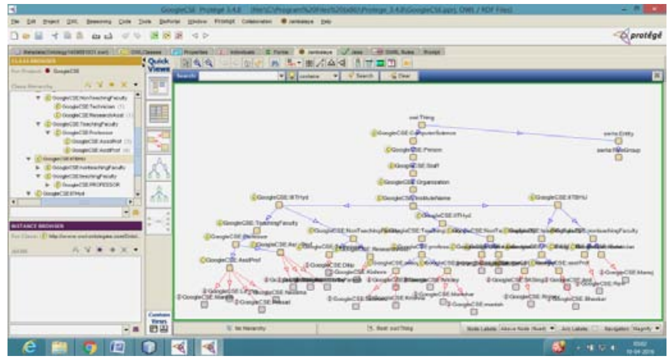

Figure 12: GoogleCSE ontology graph (source ontology)

Step 10: Comparison
It is done by comparing both ontologies using PROMPT algorithm where source ontology is “GoogleCSE.owl” and target ontology is “computerscience.owl”

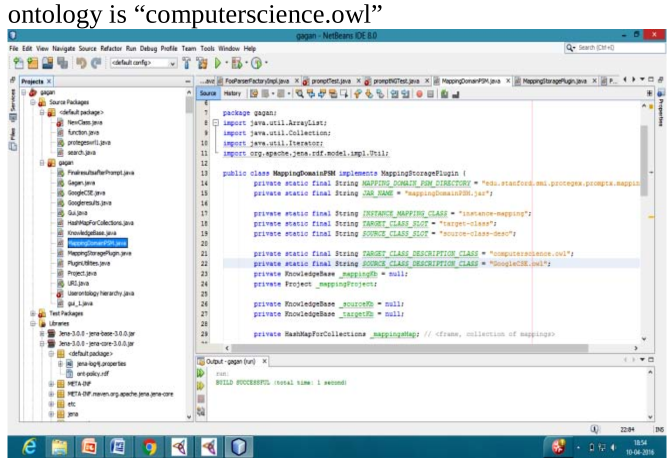

Figure 13: Execution of PROMPT algorithm

Step 11: Improved results

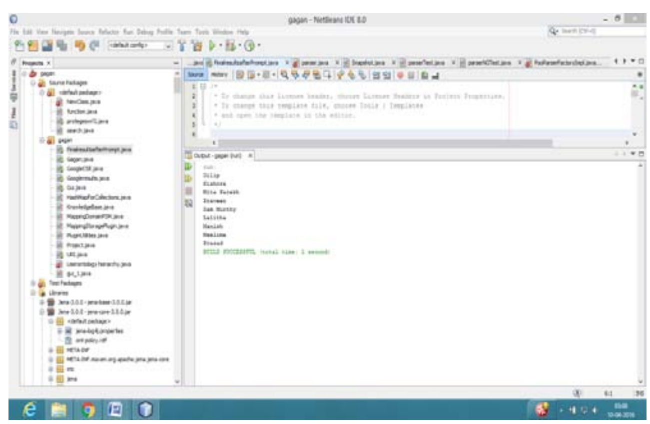

Figure 14: Enhanced Results

6.0. EVALUATION MEASURES

Sample query	Google	Our system
List the faculties of CSE in IIIT Hyderabad	Precision= 7/21 = 0.33 Recall = 7/16 = 0.43	Precision= 9/21 = 0.42 Recall = 9/16 = 0.56

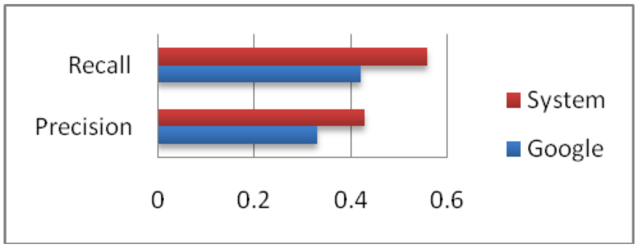

Fig 15: Higher P to R ratio of our system than GOOGLE

7.0. CONCLUSION AND FUTURE SCOPE

The given paper presents a Lexical, Ontological & Conceptual framework of Semantic Search Engine (termed LOC-SSE) with the help of semantic web technologies. The proposed system is implemented and evaluated on basis of Precision- Recall Ratio. From implementation & analysis of the proposed framework, it is concluded that given system produces more accurate results as compared to Google.

As a future work, it can be extended by developing agent based middleware search engine with the help of JADE (Java Agent Development Environment).

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Comprehensive Security Mechanism for Defending Cyber Attacks based upon Spoofing and Poisoning

Alok Pandey[1] and **Jatinderkumar R. Saini**[2]

Submitted in April, 2016; Accepted in July, 2016

Abstract – Much attention needs to be paid to different types of security threats and related attacks in the LAN and the interconnected environment. A variety of controls and counter mechanisms covering different layers of TCP/IP protocol suite are already available. But most of them have several issues related to cost, compatibility, interoperability, manageability, effectiveness etc. and hence multiple protection devices need to be installed.

In this paper we propose a comprehensive security mechanism which can detect and guard against a variety of spoofing and sniffing based cyber attacks in the local area networks. The solution does not require any additional hardware and is fully backward compatible with existing versions of ARP as no modifications are required to the existing LAN protocols. It also provides necessary detection and mitigation mechanism for the common type of DoS, MITM attacks & provides mobility along with a consistent working environment to the users as they roam around on different networks

Index Terms – ARP Spoofing, Denial-of-Service (DoS), LAN, Man-In-The-Middle (MITM), Security

1.0 INTRODUCTION & RELATED WORKS

Although a lot of effort has been made by the research community for securing the network based communication, but still there are problems which need to be resolved. More attention needs to be paid to different types of security threats and related attacks in the LAN and the interconnected environment. Malicious users can launch different types of network attacks based upon the sniffing and spoofing techniques and gather information which could be used for penetrating further into network for thefts and damages to data.

TCP/IP protocol suite was initially developed with the prime consideration of communications and as such much attention was not paid to concerned security aspects. Attackers are exploiting some of the known weaknesses of the individual protocols like ARP, ICMP, IP, TCP and UDP etc. of the TCP / IP suite.

[1]Senior Systems Manager, Birla Institute Of Technology, Mesra, Jaipur Campus, Rajasthan, India, Email Id: alokpandey1965@yahoo.co.in

[2]Professor & I/C Director, Narmada College of Computer Application Bharuch, Gujarat, India, Email Id: saini_expert@yahoo.co

Some common attacking strategies adopted by attackers are by way of Intrusions, Denial of Services [1] (DoS and DDoS), Interception and re-routing of the communication.

In a typical LAN environment, internal user can launch different types of network attacks based upon sniffing, spoofing techniques and capture sensitive information like user name, passwords, IP addresses, port numbers and other proprietary data [2] and use it for penetrating further into network for thefts and damages to data.

Capturing and analyzing a TCP/IP packet on a network for stealing network based information is called Sniffing [3]. Another well known technique for launching attacks in network environments is Spoofing[4]. This underlines the need for reliable techniques for detection of sniffing and spoofing based activities and related attacks on the network.

Attackers craft and inject bogus packets by exploiting the feature of Raw Socket Programming which is offered by most of the programming languages today. Spoofing is the process of creating and injecting fake TCP/IP packets with some one-else's identification on networks [5] whereas in MITM the entire session is hijacked by the attacker for stealing of data.

Protocols like IP and ARP are exploited [6] for launching attacks like Port Scanning, ARP Cache Poisoning, Changing of Default gateway, ICMP redirect, DHCP poisoning, DNS poisoning etc. Based upon IP spoofing, which involves forging of IP addresses of the source device, different types of attacks can be launched [7] whereas attacks like DoS and MITM can be achieved using ARP spoofing.

Address Resolution Protocol (ARP) is used for finding out the MAC address [8] of the destination device on a LAN. ARP stores such mappings of IP addresses to MAC addresses in temporary storage called cache for future usage [9]. This cache is updated from time to time. Whenever the system has to transmit a frame it first checks its ARP cache for locating the corresponding MAC address of the receiver[10]. It uses two types of messages namely ARP request and ARP reply which are encapsulated inside an Ethernet frame. It contains MAC addresses of sending and receiving devices along with a value of 0x0806 in Ethernet type [11]. The frame also contains the IP and MAC addresses of the sender and receiver along with an operation code as part of the ARP message.

The entries to the ARP cache can be added either statically or dynamically[12]. For supporting the DHCP enabled hosts, these entries are removed periodically form the cache. The devices update their ARP cache whenever they receive an ARP Reply even if they had not sent out the corresponding ARP request earlier as ARP is stateless protocol [13,14]. Thus,

despite its crucial importance ARP provides ground for launching ARP spoofing & ARP cache poisoning attacks [12].
For genuine communication both Ethernet and ARP headers should match. But since there is no mechanism to check consistency of these headers, attackers intentionally craft packets having different or forged values of IP-MAC addresses [15,16,17,18]. This is called ARP Cache Poisoning.
Thus attacker modifies the entry for gateway or any other genuine host with mapping of their IP Addresses and its MAC address in ARP Cache of victim system. After this a variety of attacks can be launched [14, 19] namely Denial of Service (DoS) attacks, Man in the Middle (MITM) attacks etc. The attackers craft different types of packets based upon IP, ICMP, TCP, UDP etc protocols and try to disrupt various functionalities of the network.
A variety of controls and counter mechanisms like Antivirus, Anti Spam, Anti Malware, Encryption and other related software covering different layers of TCP /IP protocol suite are already available. Some higher-end expensive hardware and software based devices like Switches, Firewalls, IDS, UTM etc. are also available for mitigating specific types of individual or group of attacks at various layers. But most of them do not cover the range of Sniffing, Spoofing, ARP Poisoning, Packet Crafting for Port Scanning and Flag Manipulation based DoS attacks actually taking place. Besides these, the issues related to cost, compatibility, interoperability, manageability, effectiveness etc. are also involved As a result multiple protection devices need to be installed.
Although solutions based upon Static ARP Cache entries to prevent ARP spoofing attacks exist yet they have some major issues like effort required for manual configuration of static entries, limited scalability and workability in static and DHCP based networks [14]. Some of the typical works done in this category include the DAPS (Dynamic ARP spoof Protection System- Cisco^{R}) technique suggested in [20] which is a solution to ARP spoofing that snoops DHCP packets. Katkar et al. [21] have proposed a light weight approach for prevention & detection of ARP Spoofing. A server based solution has been proposed by Ortega et .al. [22]. Another mechanism to prevent ARP spoofing based upon the use of static ARP entries was suggested by Ai-Zeng Qian[23]. A combination of using static ARP entries and SNORT-IDS is suggested in [24] for resolving the ARP spoofing problem.

2.0 METHODOLOGY

In this paper we propose a comprehensive security mechanism which can detect and guard against a variety of spoofing and sniffing based cyber attacks generated by exploiting some inbuilt vulnerabilities of heavily used major protocols of the TCP / IP suite such as ARP, IP, ICMP, TCP and UDP protocols from both the internal and external attackers.
The proposed solution performs cross layer inspection, identifies invalid combinations of Source and Destination IP addresses and MAC Addresses, performs Port Scanning, restores the Default gateways and helps the victim machine to recover from Spoofing and Poisoning based attacks in the Local Area Networks. The solution does not require any additional hardware and is fully backward compatible with existing versions of ARP as no modifications are required to the existing LAN protocols.
It also provides comprehensive detection and mitigation mechanism for the common type of Denial of Services attacks that are based upon IP, ICMP, TCP, UDP etc. protocols.
The proposed solution also aims to provide mobility and a consistent working environment to the users as they roam around on different sub networks of the corporate network which might be geographically dispersed. The proposed solution comprises of 4 Modules that work in the client server environment as follows:-
Module 1 focuses upon User Registration, Validation and Log on. Here the user is required to register and provide the required details as can be seen in Figure1. After the initial registration the user gets user name, password, an authentication code and location code which are used for logging on the network and getting the IP from the respective DHCP Server for that location [25].

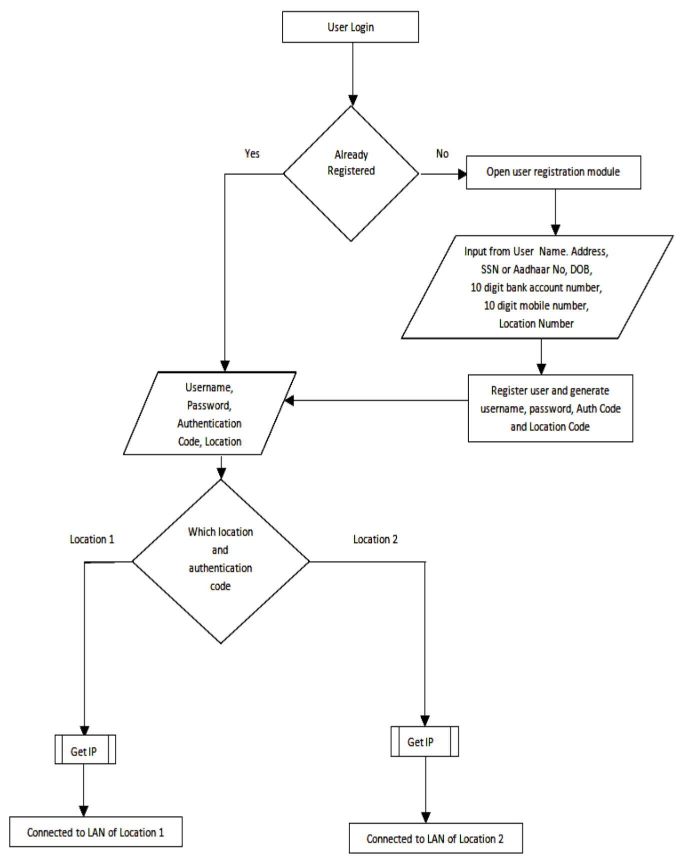

Figure 1: Client Side Flow Chart for user registration

Module 2 focuses upon the Client Side Processing. After the initial registration and verification process the client side portion takes over and provides the functionality of closing the undesired Open Ports of the system, Restoration of Default Gateway, Discovery and Reporting of the neighbors to the server and updating of the ARP Cache with the processed contents from the server. Figure2 shows process of discovering the neighbors.Another flowchart as shown in Figure3 depicts the procedures that run and scan for open ports of the system. It then selectively closes undesired open ports based upon user confirmation. It also flushes and updates the ARP Cache of the client system based upon the authenticated updates as received from the Server side from time to time.

Module 3 focuses upon Server side Processing and performs the Genuine Host Detection, Cross Layer Verification, Final Node Detection, Updating Of Client ARP Cache, Locating and eliminating attacker etc using secure data exchange as shown in flowchart in Figure4

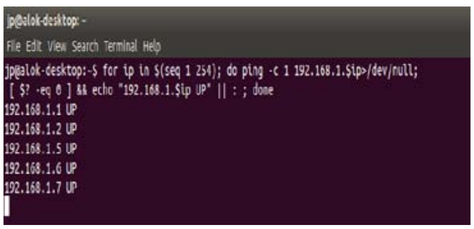

Figure 2: Client Side Scan for Active Hosts on the LAN

For detecting genuine MAC and IP Addresses of clients on the network, data regarding the active hosts on the LAN is collected using different mechanisms such as Reporting from the Clients, Lease Table of the individual location based DHCP servers for that network and Sniffing of packets on the network.

All the entries found common in all the three are recorded in the genuine host table. Other entries which are not common in all the three are recorded in the suspicious host table.

The mechanism also performs cross layer examination as seen in Figure5 and detects the invalid MAC Address – IP Address combinations by comparing the Ethernet and ARP headers [1,5,19].

If both source and/or destination MAC addresses are not identical as seen in Figure5, it means that ARP spoofing is happening on the network. The valid MAC address packets are recorded for further processing and physical node detection process.

The Final Node Detection and Cross Verification is done to verify the existence of the physical host on the network by sending ICMP ping packets to the combination of IP-MAC address pairings as recorded earlier. For the ones where no reply is received the second layer of verification is done by sending a TCP SYN packet using the details of the detected IP and MAC Address combination.

If the host is there on the LAN it will respond back with SYN / ACK or RESET packet [26]. Such entry is to be recorded in the genuine host table and passed to the client for updating its ARP Cache [27]. In case of no response then the entry is passed to the spoof alarm and counter measure mechanism for blocking the host on the network.

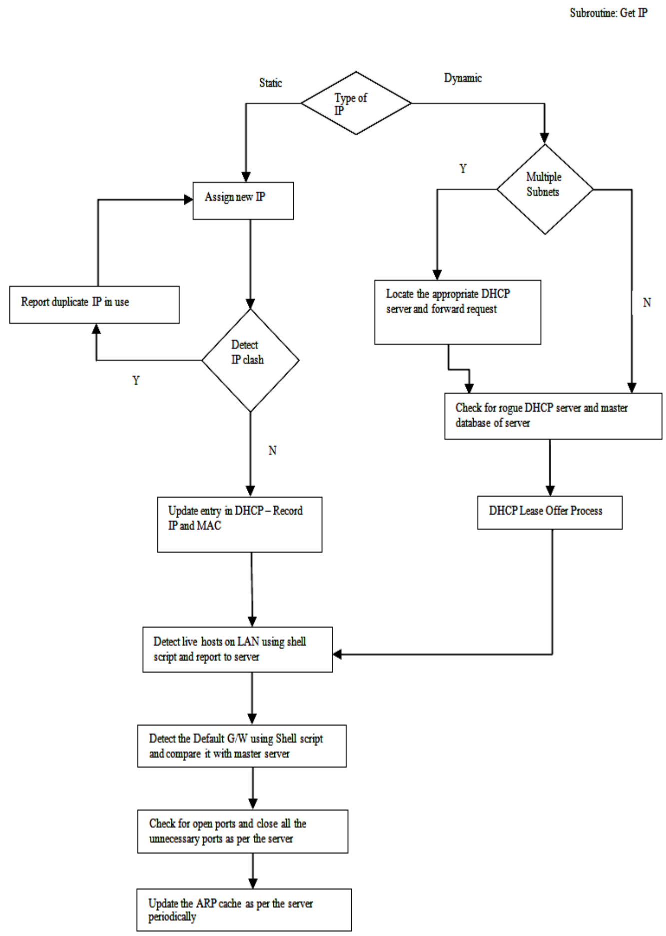

Figure 3: Client Side Flow Chart

The entries that finally pass all the tests are sent to the client for updating the respective ARP cache. The existing cache of the client is flushed out by deleting all the available entries and updating the ARP cache of individual client after a specified time period with the genuine set of valid entries as sent by server side.

Based upon the list of IP – MAC combination that failed to respond to either ICMP Ping Or SYN Scan, the administrator can eliminate such malicious hosts from the network if desired.

In order to protect the inter-host communications during the entire process communications encrypted data exchange is done between the client and server portions and also for communicating with DHCP servers on the different sub networks.

Module 4 focuses upon Prevention of Common type of DoS [31] attacks that are created using abnormal packets of TCP, UDP, IP, ICMP etc. which are purposefully crafted by attackers and injected into the networks so that the existing devices like Servers, Workstations, Computers, Firewalls, IDS and other interconnecting devices like switches, Routers, Gateways etc. and the supporting software either malfunction or crash [28]. The adversary purposefully creates the abnormality so that the filtering devices like firewalls and IDS are cheated and the packets pass through them without being blocked and ultimately result in system crashes. For the purpose of creating problems an adversary manipulates theses fields and crafts packets that are either incomplete or have wrong values.

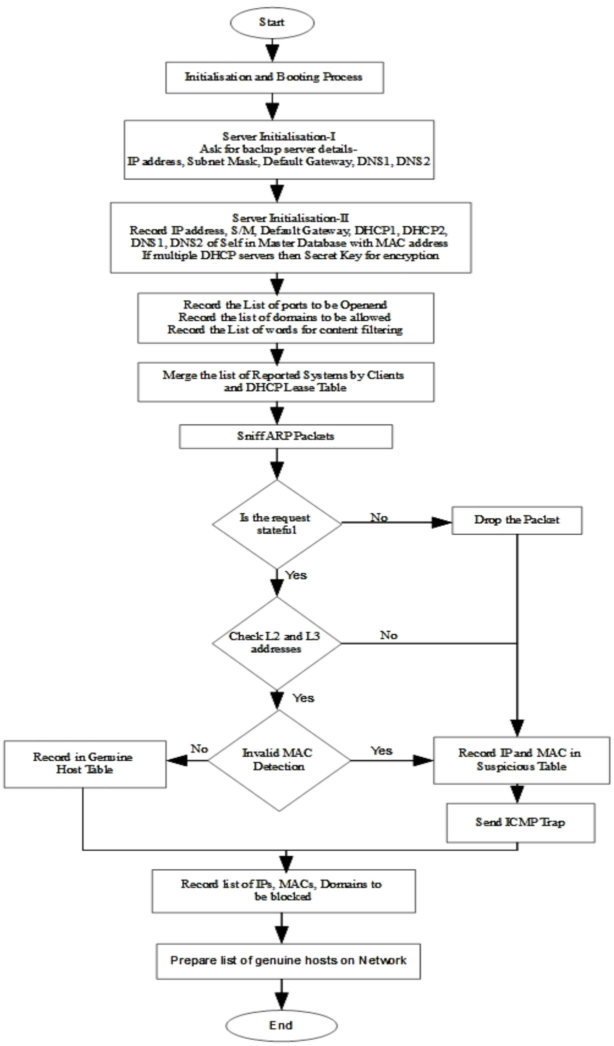

Figure 4: Server side Flow Chart

This portion filters out the abnormal packets which are deliberately crafted by the attackers to launch different types of attacks. The mechanism filters out the Abnormal IP packets which either have Unknown Protocol Type in the IP Header or are abnormally fragmented like illegal offset values, overlapping fragments, or if the first fragment is very small.

Like-wise the proposed solution is capable of filtering out the abnormal TCP Packets also. TCP uses a combination of six flags to indicate specific functionality or meaning of the current packet and its contents. Each flag is of one bit and has a specific meaning or functionality associated with it for example some TCP segments carry data while others are used for acknowledgements of the received data.

Figure 5: For cross layer examination

Out of these flags the most popular flags are the "SYN", "ACK" and "FIN", which are used for establishing connections, acknowledging and terminate connections. A SYN packet (Synchronization) is used for initiating a TCP connection whereas an ACK indicates that contents have been received and the device is ready to accept further packets. The 3-way handshake mechanism is based upon these packets and is used to ensure that both the sender and the receiver are ready to communicate before the actual transmission of data is done from either side.The detailed functional specifications for TCP are defined in RFC 793. Packets with other flag combination should be treated as suspicious. The attacker can use such illegal combination to identify the operating system at the victims system and then exploit some of its known vulnerabilities to further penetrate into the system. Sometimes such illegal combinations may go undetected through firewalls and intrusion detection systems or may crash the victim target device.

Some of the commonly seen invalid combinations for TCP may include packets with TCP Headers having both SYN and FIN Flags Set or having SYN, FIN and PSH Flags Set or SYN FIN RST Flags Set or SYN FIN RST PSH Flags Set or having only FIN Flag without ACK Flag Set or ALL Flags Set or no Flags Set at-all or SYN Flags set but containing data. Other illegal forms may include the Source or Destination Port Numbers set to 0 or packets having ACK Flag set but the Ack Number set to 0 etc. [5, 12, 17]

The proposed solution is capable of filtering the abnormally crafted UDP packets. UDP is another Protocol that is available at the transport layer. It is a connectionless protocol with very little services. It is also available in the standard deployment of the TCP IP protocol suite. Both TCP and UDP have source and destination ports. Protocols like DHCP, SNTP, DNS and TFTP use UDP as a transport mechanism [17,29,30]. The abnormal UDP packets may contain zero as either source or destination port numbers or the packet may be illegally fragmented or attacker may flood victim devices by sending multiple UDP packets with same IP address or same port numbers. [5,12,17] .

Similarly the proposed solution is also capable of filtering out the Abnormal ICMP packets, which are fragmented or redirect everything to itself or larger than 65,535 bytes or echo request packet containing data or multiple ICMP packets with same Destination IP Address [5, 12, 17, 29, 30].The communication between the client and server portions is encrypted using the encryption key which may be generated using government / authenticated individual identifications like driving license, passport number or other related information provided by the user at the time of initial registration.

3.0 EXPERIMENTAL SETUP & RESULTS

The test network consists of a network of three computers. We have used a system with two LAN cards to provide the functionality of a Router, DHCP Server and testing of the conditions. One of the machines acts as an attacker machine and is loaded with packet crafting software for generating malicious packets and injecting in the network. Wireshark was used for verification of the successful injection of such in the network..

Different types of ARP, IP, TCP, UDP and ICMP packets were crafted and injected in the network by the attacker system. Several packets with invalid source MAC Address, Destination MAC Address, Source and Destination Port Numbers, illegal flag combinations, were generated and injected in the network. Proper filter conditions to filter out such bogus packets were implemented at the Routing system.. Screenshots were taken before and after implementing the filter conditions at the victim machine. Wireshark was also run at the victim to capture and display the packets. Some of the screen shots are included herein. Figure6 shows that Packet with SYN, FIN, PSH, RST has been dropped at the Filtering System after applying the filter conditions.Figure7 shows that UDP Packet with destination port having 0 value was dropped by filter system after applying filter conditions. It can be observed that the crafted packets with illegal or invalid parameters by the attacker system are being filtered out and thus the victim system is protected against different types of DoS and MITM attacks.

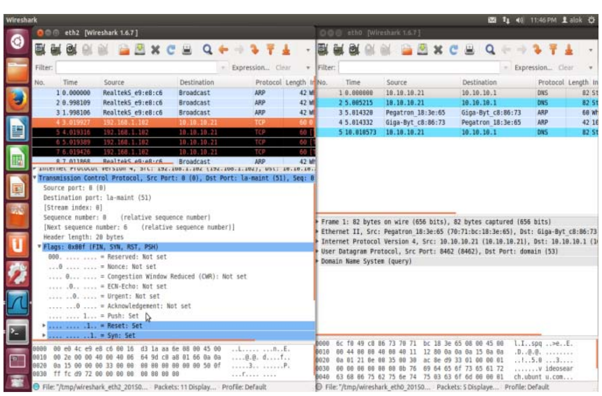

Figure 6: Packet with SYN, FIN, PSH, RST has been dropped after applying the filter conditions

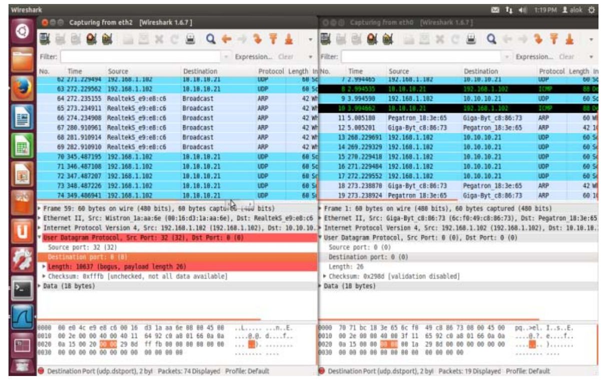

Figure 7: UDP Packet with d-port 0 value are dropped by filter system after applying filter conditions

4.0 CONCLUSION

In this paper we have highlighted the security based issues of Local Area Network and shown how some of the known vulnerabilities of the basic protocols of TCP / IP protocol suite can be exploited to launch different types of attacks. The proposed solution incorporates cross layer inspection, identifies invalid combinations of source and destination IP addresses and MAC addresses, port scanning, restoring of default gateways and helps the victim machine to recover from Spoofing and Poisoning based attacks in the Local Area Networks. The proposed solution does not require any additional hardware and is fully backward compatible with existing versions of ARP as no modifications are required to the existing LAN protocols.

The aim of this paper is purely academic research and to spread awareness amongst the network administrators and other related persons who manage, maintain and guard the networks against such attacks in LAN and WAN environments. Though highlighted, we do not intend to promote attack mechanisms nor defame any proprietary or open-source network defense tools already existing in market, we acknowledge all of them.

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State Space Model Based Channel Estimation using Extended Kalman Filter for Superposition Coded Modulation OFDM System

Rashmi N[1] and **Mrinal Sarvagya**[2]

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Abstract – In this work Extended Kalman Filter(EKF)is implemented for Orthogonal Frequency Division Multiplexing-Superposition Coded Modulation(SCM) scheme. Due to time varying nature of Rayleigh fast fading channel which vitiates the performance of data detection which in turn degrades the performance of OFDM system. An efficient channel estimation technique is necessary. An Extended Kalman filtering algorithm is proposed which is low in computational complexity. The Jakes process is modeled as Autoregressive model and approximated to Rayleigh fading channel. This estimator algorithm is bandwidth efficient and requires less computation contrast to data-based only estimators. The results obtained prove that the proposed algorithm can be used to obtain the channel estimation with low computational complexity at the receiver.

Index Terms - SCM (Superposition Coded Modulation), EKF(Extended Kalman Filter),Orthogonal Frequency Division Multiplexing(OFDM), AR model (Auto regressive)model.

1.0 INTRODUCTION

OFDM is multicarrier, bandwidth efficient modulation technique, but it suffers from Peak average Power Ratio(PAPR) and Inter carrier Interference(ICI).Due to high speed mobility of the transceiver, the channel characteristics vary rapidly. An efficient channel estimation technique is required to estimate the channel impulse response. In wireless communication system field the main efforts have been directed towards the channel estimation. Since the transmitted data are prone to the channel noise. The error controlled codes are used to minimize channel noise but these codes are bandwidth inefficient.

In order to solve these problems several coded modulation schemes are proposed in literature [1-4, 19]. To match the channel conditions both coding and modulation are combined in coded modulation schemes.

Superposition Coded Modulation (SCM) is one among the Coded modulation schemes [5-6]. The SCM system is explained in detail in section II. In this paper we are proposing channel estimation technique for the SCM –OFDM system. In most of the applications, the estimated parameters are used to detect the data transmitted.

[1]Department of Electronics and Communication Engineering,BMS Institute of Technology and Management Bangalore, India Email Id: rashmiswamy@bmsit.in
[2]Professor, Department of Electronics and Communication Engineering,Reva University Bangalore2 India. Email Id: mrinalsarvagya@gmail.com

The efficient channel estimation techniques are essential for the equalizer to work efficiently. The channel state information is very crucial in the wireless communication system. In order to maximize the SNR at the receiver, several channel approximation algorithm was proposed for rapid convergence and for improving MSE performance[7]. The channel estimation using pilot sequences were proposed in both time and frequency domains[8]. The decision direct channel estimation technique has been proposed in[9]which are based on Kalman filter. The technique mentioned in[9] has no overhead pilot symbols. But the decision direct method adopted in the[9] has delay problem. This leads to slow channel tracking in fast fading channel.

The wireless communication channel with multipath fading and Doppler effects can be approximated as a Jakes process and can be model using an Autoregressive (AR)model with white Gaussian process input. The Extended Kalman Filter (EKF) is used to extract the fast fading channel parameters [10][17].

This paper is organized as follows: section II: Describes the SCM-OFDM system. Section III:Describes Kalmanfilter based channel estimation and tracking. Section IV: Extended Kalman Filter algorithm implementation. SectionV: Simulation Results and finally SectionVI: Conclusion was made.

2.0 SCM SYSTEM

In recent years, the demand for high data rate has been increasing. Several high rate systems were proposed Superposition Coded Modulation (SCM) is one among them. In this system serial to parallel conversion of serial data stream is done, each paralleled data stream is considered as K number of data streams, each K stream is considered as individual k layer. Each k layer is encoded, interleaved and mapped. For simplicity we considered Binary Phase Shift Keying.

$$\{x_{i1}, x_{i2}, \ldots x_{ik}\}, \text{i.e., } \beta_k x_{i,k}$$

$$X_i = \sum_{k=1}^{K} \beta k x i, k \qquad (1)$$

Where each $x_{i,k}$is from a binary phase shift keying(BPSK) constellation and $\{\beta_k\}$ are a set of weighting constants[11][18].

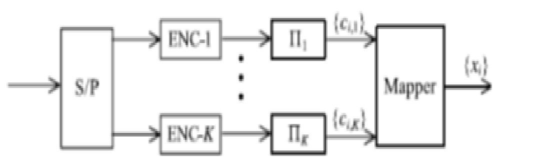

Figure 1a: Transmitter of MC-SCM

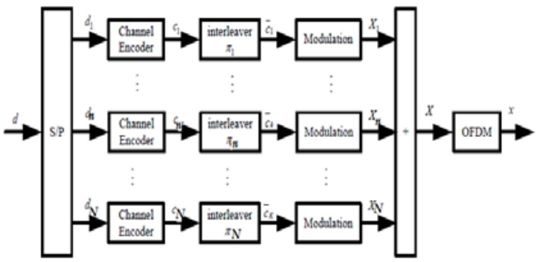

Figure 1b: Transmitter of MC-SCM-OFDM.

The output signal X is a linear superposition of N independently coded symbols. Then the superposition signal is fed into an Inverse Fast Fourier Transform (IFFT**)** modulator. After IFFT and the cyclic prefix, the signal is represented in the time domain as in equation (3)[26].

$$X(m) = \sum_{n=1}^{N} X_n(m),\ 0 \le m \le N-1 \qquad (2)$$

$$x(n) = \frac{1}{N}\sum_{m=0}^{N-1} X(m)\, e^{\frac{j2\pi mn}{N}} \quad -N_g \le n \le N-1 \qquad (3)$$

Where,
N = the total number of subcarriers

Ng = the length of CP.

The received signal $Y_n(k)$ is given by

$$Y_n(K) = \sum_{l=0}^{L-1} x_n(k-1)h_n(k-1)+w(k),\ k=0,1,\ldots N-1 \qquad (4)$$

Where $h_n(k,l)$ and w(k) represents l^{th} the complex time varying fading channel path with length L and additive noise at the k^{th} instant of SCM-OFDM symbol respectively. W (k) is assumed to be white Gaussian noise with zero mean and variance σ_n^2.
In this paper we investigate on the state space approach in modeling dynamics of the systems. In order to inference about a dynamic system and analyze the same, we require two models firstly a model which describing the state of the process with time(the system mode) and second a model relating the noisy measurements to the state (the measurement model)[21].

3.0CHANNELSTATE ESTIMATION USING EXTENDED KALMAN FILTER FOR SCM-OFDM SYSTEM.

Channel estimation is one of the major signals processing technique in wireless receivers. The channel estimation methods are classified into two types of channel estimation namely 1) Blind channel estimation: requires no training sequences, utilizes deterministic or statistical information from the received signal to estimate the channel impulse response. Advantage of this method is that it is bandwidth efficient but exhibit a draw backs such as slow convergence rate and computationally intensive 2) the second category in the channel estimation techniques to estimate the channel impulse response uses pilot symbols and previous state of the channel. The semi blind channel estimation techniques are spectral inefficient and high in computational complexity. Kalman filter attains superior estimation with few numbers of pilot symbols. Kalman filter uses an underlying channel model for channel estimation. Thus it is spectral efficient and low in computational complexity[15,16].The estimator that Kalman filter uses are of two types -channel model based estimator and data based estimator[7].

3.1 Channel model based estimator

The wireless channel is modeled as the Jakes model. Jakes model is approximated as an auto-regressive process. By solving the Yule-Walker set of equations the weights for the AR modelare determined(5). Using the relation of auto correlation'R' of the autoregressive process and the autoregressive co-efficient'r_{ss}'can be written as[15].

$$\Phi = R^{-1} r_{ss} \qquad (5)$$

Kalman algorithm uses the AR co-efficient obtained by above equations aids to model channel in a state space form [12][15].

3.2 Data based estimation

Data based estimator uses a pilot sequence of length M, transmitted with superposition coded frame. Pilot sequences are known to both transmitter and receiver. Assuming the channel is invariant along the length of the symbol.
The vector form of Pilot sequence is:

$$X=[x_0,x_1,x_2,\ldots x_{M-1}]^T \qquad (6)$$

Where 'T' represents the transpose operator.

Let the impulse response of the channel be:

$$h= [h1,h2,h_3\ldots\ldots.h_{l-1}]^T \qquad (7)$$

Where L is the process length to be tracked[13][16].
The received signal is the convolution of impulse response of the channel and transmitted pilot sequence in presence of noise as shown in equation (8). Using transmitted and received signal an estimate of the channel is found.

$$Y=X * h + n_c \qquad (8)$$

Where X is the transmitted pilot signal. Y is the received signal after passing through the channel with h as the impulse response of the channel. The n_c with zero mean and variance σ_c^2 with E_b=1 energy of the symbol, the SNR of the channel is given by:

$$\frac{Eb}{No} = \frac{1}{2\sigma c2} \tag{9}$$

The estimate of the channel is found by linear regression method as given below:

$$h=(X^TX)^{-1}(X^TY) \tag{10}$$

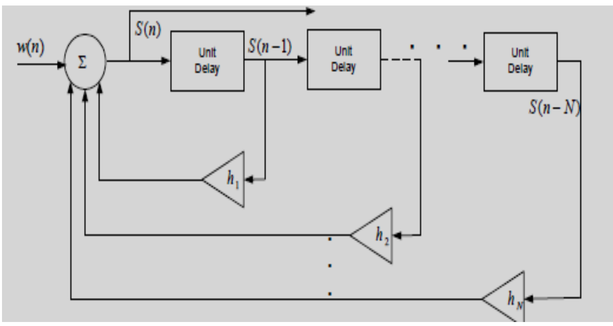

Figure 2: The Auto regressive (AR) model.

The figure represents AR process model truncated to N-taps[25].

The AR process is represented as below:

$$S(n)= \sum_{i=1}^{N} \emptyset i S(n-i) + w(n) \tag{11}$$

S (n): The complex Gaussian process.
φ_i: AR modelparameters.
N=the number of taps delays.
W(n): Zero mean Complex Gaussian random variables.
The N^{th} order difference equation in the form of vector for state space model is written as:

$$\bar{S}(n) = F\bar{S}(n-1) + \bar{W}(n) \tag{12}$$

Where $\bar{S} and\ \bar{W}$ are column matrix of size (N X 1) and F is an (NXN) matrix [16][20].The mean and variance of autoregressive process are zero and

$$\text{Variance}= \sum_{i=1}^{N} \emptyset i Rss(i)+\sigma_n^2 \tag{13}$$

Autocorrelation is given by

$$\begin{aligned} R_{ss}(m) &= E\{S(n-m)S(n)\} \\ &= E\{[\textstyle\sum_{i=1}^{N} \emptyset i S(n-i) + w(n)]S(n-m)\} \\ &= \textstyle\sum_{i=1}^{N} \emptyset i Rss(m-i) \end{aligned} \tag{14}$$

The autocorrelation co-efficient is:

$$rss(m)=\frac{Rss(m)}{S2x} = \sum_{i=1}^{N} \emptyset irss(m-i), \quad m\geq 1 \tag{15}$$

The above equation in matrix form is called Yule-Walker equation.

$$\bar{R}\emptyset = \bar{r}ss \tag{16}$$

Since Ris invertible, we obtain

$$\emptyset = \bar{R}^{-1}\bar{r}ss \tag{17}$$

The equation (17) can be used to express the underlying process model which in turn can be used to express a Kalman filter to estimate the process.[13][16].

3.3 System model:

The system model for the first order process is given by:

$$S(n)=\varphi_i S(n-1)+w(n) \tag{18}$$

S (n):The complex Gaussian process.
φ_i:AR modelparameters.
W(n): Sequences of i.i.dcomplex Gaussian random variables withzero meanwith variance σ_w^2[16]

3.4 Observation model

The data aided estimate creates estimates of the noisy version of the Complexprocess[16].
The observation model of the estimate can be written as:

$$X(n)=S(n)+v(n) \tag{19}$$

Where S(n)= The complex Gaussian process at time n.
X(n)=The data based estimate of S(n)
V(n)=Error of the data based estimate[16,20].

Algorithm to track the process is as follows:

Step 1: The initial conditions are:

$$\hat{S}(0) = E\{S(n)\} = 0$$

$$P(1) \geq \{\sigma_w^2 \text{ and } \sigma_v^2\}$$

Step 2: The Kalman gain is given by:

$$K(n) = \frac{P(n)}{P(n)+\sigma v2(n)}$$

Step 4: The current estimate of the process, after receiving the data estimate is given by:

$$\hat{S}\ curr(n) = \hat{S}(n) + k(n)[X(n) - \hat{S}(n)]$$

Step5: the predict estimate of the process is given by:

$$\hat{S}(n+1) = \emptyset 1\{\hat{S}\ curr(n)\}$$

Step6: the current error co-variance is given by:

$$P_{curr}(n)=[1-k(n)]P(n)$$

Step7: the prediction error covariance is given by:

$$P(n+1)=\emptyset_1^2\{P_{curr}(n)\}+\sigma_w^2$$

4.0 THE ITERATIVE EXTENDED KALMAN FILTER IMPLEMENTATION

The simulation setup parameters are as follows:
System equation:

$$S(n)=0.9S(n-1)+w(n) \tag{20}$$

Observation equation:

$$X(n)=S(n)+v(n) \tag{21}$$

M=8: The pilot training sequence length.
The SNR of the channel for unit energy, E_B=1 is:

$$\frac{EB}{No}=\frac{EB}{2\sigma c2}=6\text{dB}$$

Thus σ_c^2=0.1256

The data estimate variance

$$s_v^2=\frac{\sigma c2}{M}=0.0157$$

The variance of the noise is

$$s_w^2 = 2\sigma_v^2=0.0314$$

Channel estimation algorithm using theKalman filter is as follows:
{Initialize:
Time:n=1

Assuming initial predicted error covariance and initial prediction as P(1)=1,$\hat{S}(1)$=0; respectively.
{Start iteration:
Perform data based estimation, i.e. get X(n)[15,16,22].

Calculate Kalman gain:

$$K(n) = \frac{P(n)}{P(n)+1}$$

Calculate current estimate:

$$\hat{S}curr(n) = \hat{S}(n) + k(n)[X(n) - \hat{S}(n)]$$

Calculate current error covariance:

$$P_{curr}(n) = [1-k(n)]P(n)$$

Predict ahead:

$$\hat{S}(n+1)=0.9\{\hat{S}curr(n)\}$$

Predict the error covariance:

$$P(n+1) = (0.9)^2\{[1-k(n)]P(n)\}+1$$

Time :(n=n+1)
End loop}

5.0 SIMULATION RESULTS

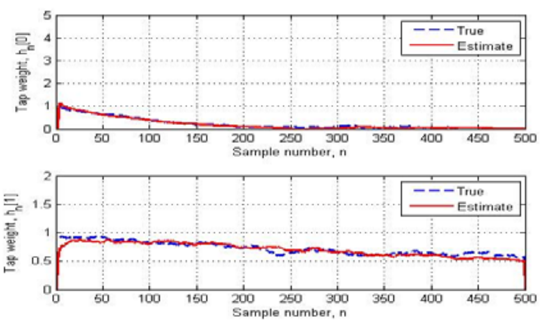

Figure 3: Shows channel estimation for time varying channelwith different tap gain.

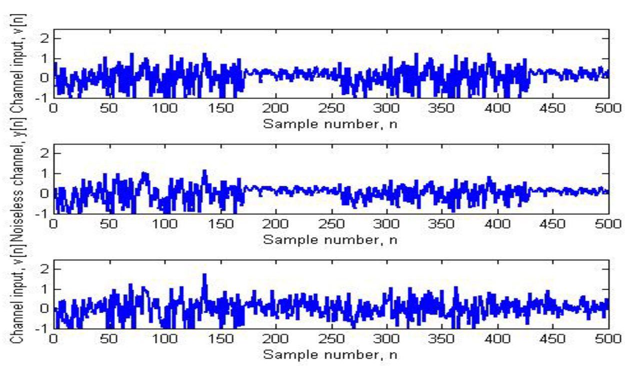

Figure 4: Input and output of the channel

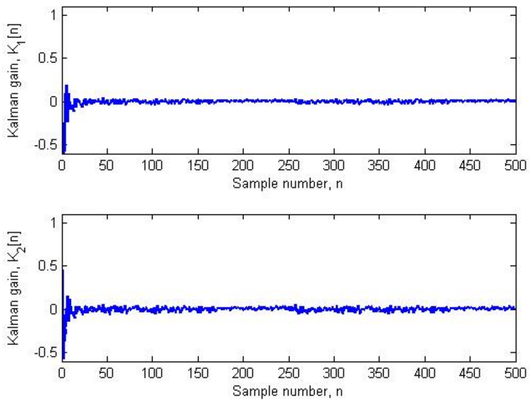

Figure 5: Kalmangain for different sample values

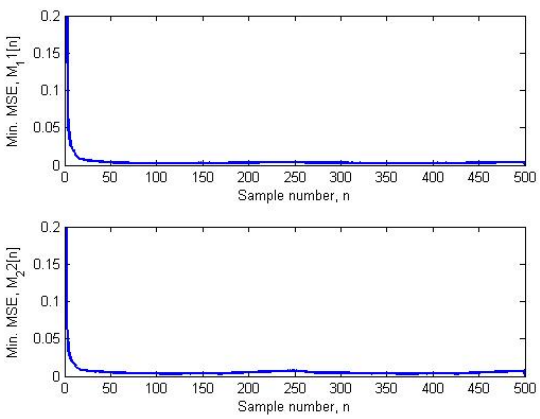

Figure 6: Variations of errors with iterations.

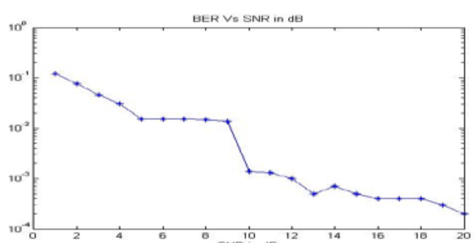

Figure 7: Bit Error Rate VsSNR(dB)

The Extended Kalman filter algorithm has been implemented for SCM-OFDMsystem with pilot sequence of 8 bits is inserted in each data block containing 200 bits with 100 iterations. Figure 4 shows the input of the channel without noise and including noise. Figure 5 shows the channel tracking capability of Extended Kalman filter for the SCM-OFDM system time varying channel. Figure 6shows Kalman gain for channel estimation using Extended Kalman filter used to estimate the SCM system with Gauss Markov channel.

Figure 7 shows MMSE variation obtained by taking actual and estimated complex tap gain using the Extended Kalman filter.

6.0 CONCLUSION

Extended Kalman filter has been implemented for the Superposition coded modulation system with OFDM systems. With the minimum length of pilot sequenceand less number of iterations,Extended Kalman filter algorithm can track the channel.

Thus it proves that EKF is a bandwidth and low complexity algorithm for SCM-OFDM system. With fewer numbers of iterations AR model demonstrates an optimum estimate of the channel impulse response. The performance of thealgorithm has been improved further by using MMSE analysis after the estimate is obtained.

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An Emotional Model based on Wavelet Coherence Analysis of EEG Recordings

Anas Fattouh[1,2]

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Abstract – Molding emotion from electroencephalography (EEG) recordings has been always a challenge to researchers. They have developed some models by analyzing the temporal and/or spectral properties of EEG recording. However, analyzing the temporal and spectral synchronization between two EEG recordings can provide information about the functional connectivity of the brain undetectable by the conventional methods. This paper presents a wavelet coherence analysis of EEG recordings acquired from healthy subjects while they are listening to verses from the Quran recited by different reciters. The results showed that the connectivity between some pairwise electrodes changes significantly with the emotion reported by the subjects. Therefore, they can be used as a model to estimate the emotion.

Index Terms – Auditory Stimuli, Brain Computer Interface (BCI), Electroencephalography (EEG), Emotion, Emotiv EPOC Neuroheadset, Functional Connectivity, Wavelet Coherence.

NOMENCLATURE

Brain Computer Interface – BCI, Coherence Slope – CS, Continuous Wavelet Transform – CWT, Cross-Wavelet Transform – XWT, Discrete Wavelet Transform – DWT, Electroencephalogram – EEG, Fast Fourier Transform – FFT, Frequency Averaged Wavelet Coherence – FA, Kernel Density Estimation – KDE, Power Spectral Density – PSD, Wavelet Coherence – WC, Wavelet Transform – WT

1.0 INTRODUCTION

Emotion plays an important role in our thinking and behavior and therefore understanding and estimating emotion can improve the communication between humans, from one side, and between humans and machines, from another side. Emotion can be recognized from external body's signals such as text, facial expressions, speech, body's gestures, or a combination of several signals [1-5]. Recently, body's internal signals are used to discriminate between emotions such as heart rate and brainwaves [6-7].Emotion's recognition has many applications in different domains such as education, heath, commerce, games, security, and many others [8-15].

However, the most important application for the computer scientists could be the natural language processing domain where the machine can understand the emotion of the user and react upon this understanding [16-17].

Recognizing emotion from brainwaves is based on identifying distinct EEG patterns associated with different emotions. It is usually done by analyzing the EEG recordings in time domain, frequency domain, or both. Correlation functions are used in time domain to identify similarities between segments from one recording or between two recordings [18]. Frequency domain reveals information that is invisible in time domain. Fast Fourier transform (FFT) is used to convert the recordings from the time domain to the frequency domain. Then the power spectral densities (PSDs) of the converted signal are calculated in several frequency bands and compared to find any possible association with the studied phenomena [19]. Other methods were proposed to estimate the power spectral densities (PSDs) such as the kernel density estimation (KDE) method [20].

Recently, wavelet transform (WT), both continuous wavelet transform (CWT) and discrete wavelet transform (DWT), was successfully used in many applications to find the oscillation in signals in time-frequency domain [21-23]. FFT and WT are similar transforms in the sense that they both measure the similarity between a signal and an analyzing function, however, they differ in their choice of analyzing function.

While the wavelet transform (WT) develops a time series into a time-frequency space, cross wavelet transform (XWT) can be used to find regions in time-frequency space where the two time series show high common power. Moreover, the wavelet coherence (WC) can be used to find regions in time frequency space where the two time series co-vary, but does not necessarily have a high power [24-25].

In this paper, the wavelet coherence (WC) is computed for selected pairs of the adjacent EEG recordings acquired from healthy subjects while they are listening to verses from the Quran recited by different reciters. The results showed that some pairwise recordings have high covariance values, while others have low covariance values. Using these covariance values, a connectivity map of the brain can be produced. The connectivity map showed the regions in the brain that are correlated with the emotion. Furthermore, the results showed that the covariance values of some adjacent pairwise recordings vary approximately linearly with the emotion. The slope of the approximated linear map, called the coherence slope (CS), is computed by fitting a first-degree polynomial function to the covariance values. This approximated linear map is considered as an emotional model that can be used to estimate the emotion from the covariance values of selected pairwise electrodes.

[1]Department of Computer Science, Faculty of Computing and Information Technology, King Abdulaziz University, Jeddah 21589, Saudi Arabia, email: afattouh@kau.edu.sa.
[2]Department of Automatic Control and Automation, Faculty of Electrical and Electronic Engineering, Aleppo University, Syria, email: fattouh@gmail.com.

2.0 METHODOLOGY

The process of building the proposed emotional model passes through two phases, the data collection phase, and the data analyzing phase. The data collection phase is an online phase where a designed experiment is run and the EEG data is recorded. The data-analyzing phase is an offline phase where a first-degree polynomial function is fitted to the data. These steps are explained in detail in the following subsections.

2.1 The Subjects

Seventeen male volunteers with age in the range 16 years to 45 years participated in the experiment. Participants were from different nationalities and had no past psychiatric or neurological disease. They also have no experience in brain computer interface (BCI) experiment. They have to sign an informed consent prepared in accordance with the regulation of the local ethic committee.

2.2The Stimuli

Seventy-Five verses recited by five different reciters were selected by experts based on the meaning of these verses. They expect that they could evoke different emotions. Ten seconds of a blank is added to the beginning of each verse to be used as a baseline for the recorded data.

2.3EEG Recording

EEG recording passes through three units, signal acquisition unit, importer unit and recording unit. The signal acquisition unit used in the proposed system is the Emotiv EPOC neuroheadset shown in Fig.1. The properties of theEmotiv EPOCare given in Table 1. The acquired signals are aligned, band-pass filtered, and digitized at frequency 128 Hz and wirelessly transmitted to a windows PC [26]. The importer unit is a Matlab®-based server that streams EEG signals acquired by the Emotiv headset to a Simulink® model in real-time [27]. The imported signal is recorded from inside Matlab® using a dedicated function [28].

Figure 1: The recording device (Emotivneuroheadset)

2.4The Experiment

The experiment starts with a pre-session, where the subject is informed about the experiment and the steps to follow in order to complete the experiment successfully; then consent is signed by the subject. The experiment is performed in a calm room with low lighting and comfortable ambient. The subject sits on an armchair in front of a PC. The Emotiv headset is mounted on the subject's head and the data acquisition program starts on the PC.

Table 1: Properties of the Emotiv EPOC

Number of channels	14 (plus CMS/DRL references, P3/P4 locations)
Channel names (International 10-20 locations)	AF3, F7, F3, FC5, T7, P7, O1, O2, P8, T8, FC6, F4, F8, AF4
Sampling method	Sequential sampling. Single ADC
Sampling rate	128 SPS (2048 Hz internal)
Resolution	14 bits 1 LSB = 0.51μV (16 bit ADC, 2 bits instrumental noise floor discarded)
Bandwidth	0.2 - 45Hz, digital notch filters at 50Hz and 60Hz
Filtering	Built in digital 5th order Sinc filter
Dynamic range (input referred)	8400μV (pp)
Coupling mode	AC coupled
Connectivity	Proprietary wireless, 2.4GHz band
Power	LiPoly
Battery life (typical)	12 hours
Impedance Measurement	Real-time contact quality using patented system

After ensuring that the Emotiv electrodes are well connected with the program, the subject selects a verse and starts listing to it while he is focusing on the meaning of it. After each recording, the subject reports his emotion using on a scale between 0 and 100 where 0 means completely unhappy and 100 means completely. The experiment procedure is shown in Fig. 2.

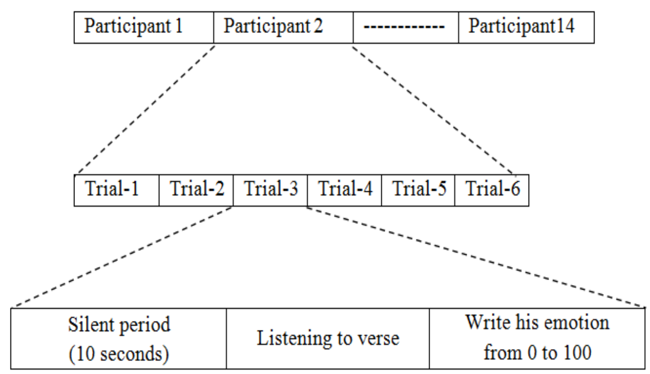

Figure 2: Procedure of the experiment

2.5Data Pre-Processing

The raw EEG signal is typically quite noisy; therefore, it is necessary to clean it up. To this end, a Butterworth bandpass filter of order eight in the band 2 to 42 Hz is used. Then, the average of all channels is used to references the filtered EEG signals by subtracting it from each channel. The last step is removing a baseline of the EEG signal from each channel for all recordings.

2.6 Pairwise Recordings

In this work, the EEG data are recorded from fourteen electrodes in the 10-20 International System as shown in Fig. 3. As the instant wavelet coherence (WC) is considered in this study, only the local electrode pairs in the frontal lobe are selected to avoid errors resulted from the propagation delays and the other electrical effects. Local electrodes pairs are defined as neighbor electrodes on the scalp. Each electrode has 3 to 6 adjacent electrodes as shown in Table 2.

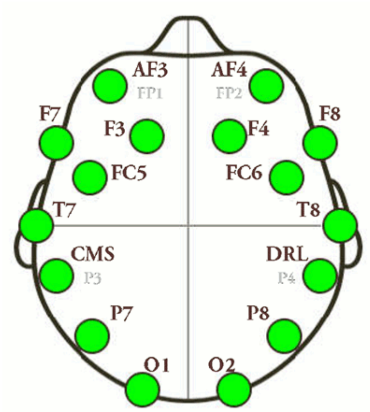

Figure 3: Fourteen electroencephalogram (EEG) electrodes in theEmotivneuroheadset

Table 2: Selected Adjacent Electrodes

Elect-rode	Adj. 1	Adj. 2	Adj. 3	Adj. 4	Adj. 5	Adj. 6
AF3	AF4	F3	F4	F7	FC5	
AF4	AF3	F3	F4	F8	FC6	
F3	AF3	AF4	F4	F7	FC5	FC6
F4	AF3	AF4	F3	F8	FC5	FC6
F7	AF3	F3	FC5			
F8	AF4	F4	FC6			
FC5	AF3	F3	F4	F7		
FC6	AF4	F3	F4	F8		

2.7 Wavelet Coherence

EEG recordings are nonstationary time series, which means that their frequency content changes over the time. In order to detect their common time localized oscillations, it is necessary to measure their coherence in the time-frequency domain. Wavelet coherence is an appropriate tool for this purpose and it will be used in this study. The computing formula of the wavelet coherence (WC) can be found in [24-25].

2.8Proposed Emotional Model

Consider two time series $x(t)$and $y(t)$ coming from two adjacent electrodes and assume that their wavelet coherence is $WC_{xy}(t,f)$. Then, their coherence slope CS_{xy} is the slope of the line fitting the frequencyaveraged wavelet coherence $FA_{xy}(t)$defined as:

$$FA_{xy}(t) = \frac{1}{f_2 - f_1} \int_{f_1}^{f_2} WC_{xy}(t,f)\,df \quad (1)$$

wheref_1 and f_2 are the lower and upper frequencies of $WC_{xy}(t,f)$. Therefore, the relationship between the emotion and the frequency averaged wavelet coherence $FA_{xy}(t)$can be approximated by the following linear map:

$$FA_{am}(k) = CS_m Emotion(k) + c \quad (2)$$

Where$CS_m$is the maximum coherence slope between all pairs of selected adjacent electrodes (see Table 2), FA_{am} is the time average of $FA_{xy}(t)$ for the pair with maximum coherence slope, and c is a constant.

3.0 RESULTS AND DISCUSSION

Fig. 4 shows the verses and the reciters selected by the 14 subjects performed the experiment (the blue and green circles respectively). It also shows the emotion reported by the subject after each experiment (the red circles).From Fig. 4, one can see that the same verses recited by the same reciter produce different emotions for different subjects. For example, verse no. 1 from reciter no. 6 produces emotion 50 for subject 1 and 80 for subject 5.

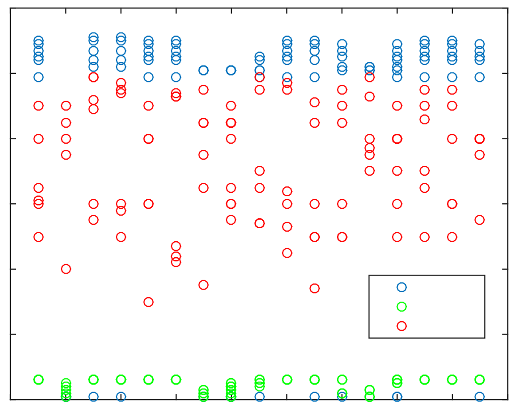

Figure 4: Experiment data

Fig. 5 shows a segment of the EEG recordings from one subject acquired from the14 Emotiv headset electrodes during the execution of the experiment.

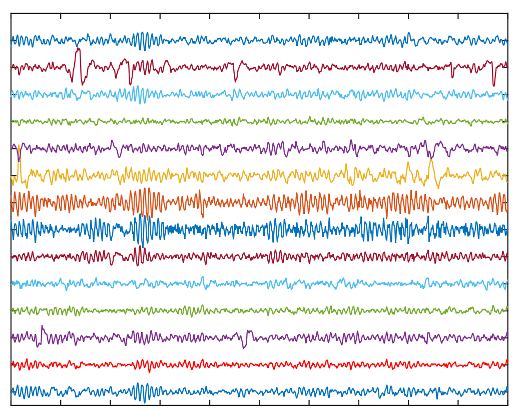

Figure 5: Segment of EEG recordings from a subject

Wavelet coherence is computed for each selected pair of adjacent electrodes according to Table 2 and for each subject [29]. Fig. 6 shows the wavelet coherence of the pair AF3-F7 from the EEG recordings of one subject. In Fig. 6, the horizontal axis shows the time while the vertical axis shows the frequency (the lower the frequency, the higher the scale).

Yellow regions in time-frequency domain represent significant interrelation between the two recordings. The arrows show lead/lag phase relations between the two recordings. When the two recordings move in the same direction, they are in phase and the arrows point to the right. When they move in the opposite direction, they are in anti-phase and the arrows point to the left. Arrows pointing to the right-down or left-up indicate that the first recording is leading, while arrows pointing to the right-up or left-down show that the second recording is leading [29].

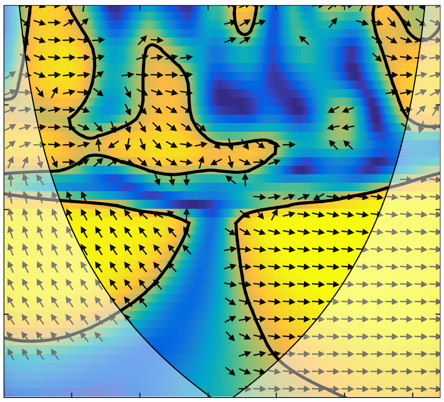

Figure 6: Wavelet coherence (WC) of AF3-F7 from the EEG recordings shown in Fig. 5

The wavelet coherence for each pair of adjacent electrodes and for each subject is averaged over the frequency domain to the get a new time series called FAxy time series. Fig. 7 shows a sample of these FAxy time series. For the time seriesFAxy obtained from the same pair of electrodes and from all subjects, a linear polynomial function is fitted to its averaged values over the time.

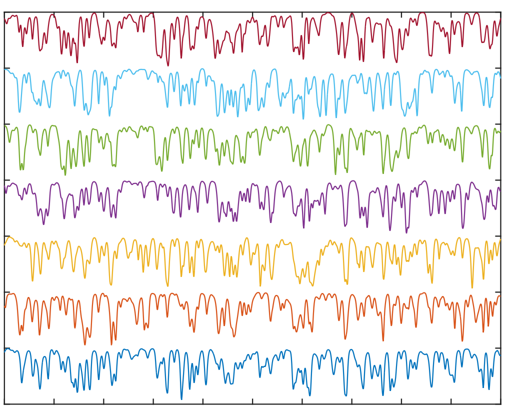

Figure 7: Frequency averaged wavelet coherence (FA_{xy}) of some EEG recordings

Fig. 8 shows the values of an averaged FAxy time series and a linear polynomial fitted function to the values.The slope of the fitted linear function is calculated for each of considered averaged FAxy time series. Table 3 shows these slopes (CSs) for all considered pairs of electrodes and for all subjects.

From Table 3, we can see that some pairs of electrodes have high positive CS values, which means that they are correlated to the emotion, and they can be used to estimate the emotion.

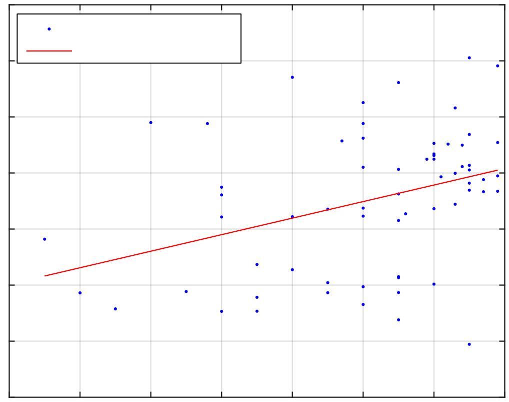

Figure 8: Average FA_{xy} versus emotion of the pair F7-FC5

Table 3: Coherence Slope (CS) of Selected Pairs

Pair Name	CS%	Pair Name	CS%
F7-FC5	2.288	F7-FC6	0.5304
AF3-F7	2.157	AF4-F3	0.4484
AF3-FC5	1.621	AF3-F8	0.3393
AF3-AF4	1.489	F3-F4	0.3139
F3-FC5	1.479	AF3-FC6	0.05809
AF4-FC5	1.42	F3-FC6	0.02811
F4-FC5	1.369	AF3-F3	0.01177
AF4-F7	1.363	F4-FC6	-0.02839
F3-F7	1.269	AF4-F4	-0.1085
F4-F7	1.23	F3-F8	-0.1662
F8-FC5	0.8892	AF4-F8	-0.1948
FC5-FC6	0.8515	AF4-FC6	-0.2343
F7-F8	0.6965	F4-F8	-0.4114
AF3-F4	0.5337	F8-FC6	-0.6889

4.0 CONCLUSION AND FUTURE WORK

In this work, a wavelet coherence analysis is performed on EEG recordings acquired from 14 healthy subjects while they are listening to verses from the Quran recited by different reciters. The results showed that some pairs of electrodes exhibit more correlation with the emotions reported by the subjects. More specificity, the results showed that the pair AF3-F7 possess the higher correlation value and it could be used to estimate the subject's emotion from its wavelet coherence value.

As a future work, the proposed emotional model will be evaluated in real time application. Moreover, other averaging and fitting methods could be considered.

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VANET: Expected Delay Analysis for Location Aided Routing (LAR) Protocol

Kamlesh Rana[1], Sachin Tripathi[2] and **Ram Shringar Raw[3]**

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Abstract – The Vehicular Ad-hoc Network (VANET) is a networking technology uses to create wireless network using mobile vehicles. In this network, the mobile vehicles work as intermediate node to transmit data packets over the wireless network and mobile nodes are free to move from one network to other network. All nodes in VANETs are highly movable and their movements area and direction to be restricted within a predefined geographical areas. All nodes in VANETs nodes must has to follow certain conditions such as their speed to be restricted by standard speed limit, patterns of the road, and traffic conditions. Due to high speed of nodes in the network, links may breaks frequently due to this failure in data delivery occurs in the network. Routing of data packet in the VANETs are more challenging task due to highly dynamic natures. Performance of the VANET depends upon the selection of the suitable next-hop nodes for further transmission in the network. Due to highly movable vehicular nodes, the link may break very frequently caused network delay; therefore, it is important to select the appropriate next-hop forwarding node to complete the data transmission. In this paper, we have computed the node distribution at the border area, expected one hop distance and expected delay using one-hop distance between first two nodes. Further, we have analyzed the numerical results using MATLAB.

Index-Terms - VANET, Expected Distance, IVC, V2V, LAR, Expected Delay.

1.0 INTRODUCTION

VANET is a wireless local area network uses vehicles as mobile nodes to form an ad hoc network. In VANET, efficient routing protocols are needed to transmit data packets over the network via number of intermediate nodes. In order to this, if intended destination vehicle is inside the communication range of the source vehicle, then source uses unicast forwarding and directly send message to the destination node. If destination node is out of the communication range of the source's node, then source node use multicast forwarding technique to send message [4].

VANET integrates wireless networks capabilities and have a number of applications such that traffic monitoring, traffic control, blind crossing, collisions preventing, and services of near by information, and real time detour routes and computation. VANET enables an Intelligent Transport System (ITS) known as Inter-Vehicle Communications (IVC) or Car -to-Car (C2C) communications that is used for vehicle-to-vehicle communication.

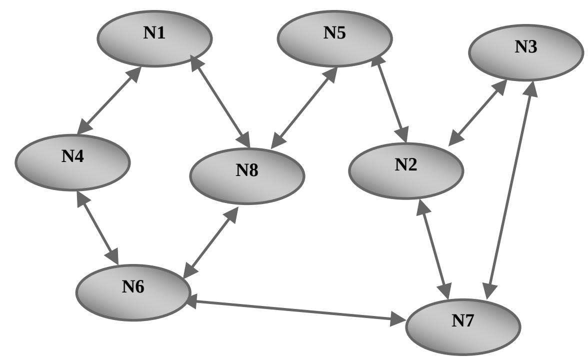

Figure 1: Movement of Mobile Nodes in Ad-hoc Network

To transmit data packets throughout the network, if destination node is out of the transmission range of the sender node some routing protocols to be required. The basic aim of routing protocol is to search a better route for packet delivery to the correct destination in the network. Now these day study of various routing protocols in VANET have been a popular research area for researchers from many years. According to behavior of the routing protocol, we can classify it into three types such as Reactive, Proactive and Hybrid routing protocol.

In Reactive routing protocols whenever a node would like to communicate with other nodes on the network. Then communicating node initiates a RREQ message with other nodes throughout the network. Once a route discovery process has been finished the route maintenance process has to maintain the route until the destination node becomes inaccessible [7].

Functionality of Proactive routing protocol is that each node maintains the location information of it's neighbors in the network. If there is any change in the current location of any node in the network the routing table will be updated this information because routing information is time to time periodically transmitted throughout the network, The Proactive routing protocol has a measure advantage such as the transmission will be completed without any delay, if a route already existing the network before arriving of traffic in the network. If route already does not exist in the network the data packets has to wait until unless node receives routing information of corresponding destination node. The proactive

[1,2]Department of Computer Science and Engineering, Indian Institute of Technology (ISM), Dhanbad, Jharkhand, India,
[3]Department of Computer Science and Engineering, Indira Gandhi National Tribal University, M.P., India,
[1]ranakamles@gmail.com, [2]var_1285@yahoo.com
[3]rsrao08@yahoo.in

routing protocols requires significant amount of resources for keeping routing information up-to-date and reliable for highly dynamic network topology[9].

The hybrid routing protocol combinations advantages of the both routing protocols such as proactive and reactive. Initially proactive routing protocol is responsible to establish a connection after this reactive routing protocol takes care of remaining operations. The main disadvantages of this routing protocol are:

- **i.** Depends upon the number of other nodes activated.
- **ii.** Traffic demand reaction depends on gradient of traffic volume.

Zone Routing Protocol (ZRP) and Zone Base Hierarchical Link State Routing Protocol (ZHLS) are examples of hybrid routing protocols. ZRP uses IARP as pro-active and IERP as reactive component.

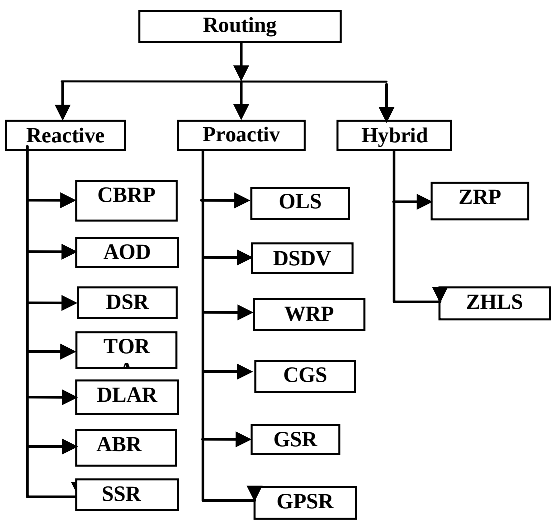

Figure 2: Classification of Routing Protocols in VANET

Generally, VANET has following three types of communications:

- **i.** C2C Communication
- **ii.** C2I Communication
- **iii.** C2R Communication

1.1. C2C (Car to Car) Communication

The basic purpose of Car to Car communication is to increase driving comforts and safety measures during driving time. C2C communications to be integrated with a wireless network with the help of those automobile nodes send messages to each other. The data transmitted on the network must have to include certain parameters such as speed location and of node, travelling direction of node, braking, and loss of stability. The C2C communication uses a technology known as DSRC (dedicated short range communications). The C2C communication uses one of possible frequency such as 5.9 GHz so it can be referred as Wi-Fi network. The Range of this varies from 300 meter to 1000 feet or about 10 seconds at a highway. The Car-to-Car communication behaves like a mesh network because every node in this can send, retransmit and capture signals. With 5 to 10 nodes technology in the network is able to gather a mile a head traffic conditions.

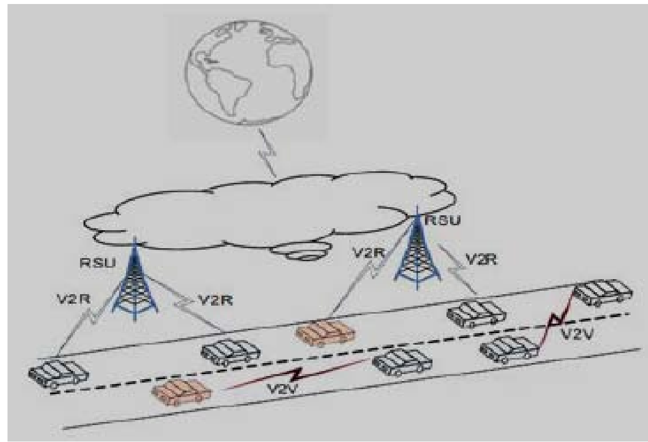

Figure 3: Various communications through VANET[21]

In Car-to-Car communication car driver must be able to receive an alert message such as a flashing red light in the instrument panel. Most of the prototypes have been advanced to stage where the cars brake. Sometimes steer around hazards C2C could captured and transmit these inputs, among others. C2C arrives in cars by time to time; some may be stripped out for the sake of simplicity or cost-cutting.

- Velocity of vehicle
- Position of vehicle and direction of travel
- Accelerating and slowing
- Brakes on and anti-lock braking
- Lanes Changing
- Controlling of stability
- Windshield wipers, defroster and headlamps should be on in daytime in case of raining and snowing
- Gear positioning

1.2. Car to Infrastructure (C2I) Communication

Infrastructure in the Car to Infrastructure Communication works as a coordinator between devices; it gathers information about the traffic and road conditions globally or locally. Then after, certain behaviors to be imposed on a group of vehicles. The ramp metering is an example of Car to Infrastructure communication that requires limited sensors and actuators such as traffic density measurements and traffic light on the highway.

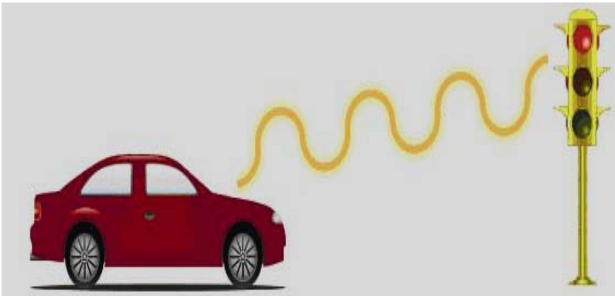

Figure 4: Car to Infrastructure Communication [22]

The infrastructure in the Car to Infrastructure Communication suggests the velocities and accelerations of vehicle and inter-vehicular distance on the basis of the traffic conditions if scenario is more sophisticated. The main objective of this is to optimize overall emissions, fuel consumptions and traffic velocity. If there is any suggestion for a vehicle it could be broadcast to the drivers via wireless connections such as via road displays or directly to vehicles.

Further if there is any suggestion it could be integrated into the vehicle controls and implemented semi-automatically. For longer-range vehicular networks Car to Infrastructure communication widely provides solutions for the problems. Preexisting network infrastructures are used by C2C communication such as wireless access points such as Road-Side Units (RSUs). Car to infrastructure (C2I) signs and signals could transmit traffic and weather indicators:

- Green, yellow and red are traffic signal phase.
- Stop sign.
- Left turn should not be at the left turn.
- Temperature at a bridge that freezes over before the ground
- Cars ahead signals
- Approaching emergency vehicle

To get true picture of traffic scenario of ahead multiple sensors are used on the multiple cars. Suppose that, if only one car's wiper is on, then it will indicate that the driver is cleaning the windshield or stalk is hilted by the mistake; if a large number of cars are doing the same things then it will show raining or snowing. If a tight cluster does, it could be a truck spewing oil or snowmelt dripping off a bridge. That could be verified by the average speed of all the cars.

1.3 Car to Roadside (C2R) Communication

In Car to Roadside Communication, the Roadside Unit (RSU) acts as wireless access vehicular environment (WAVE) providers. The WAVE keeps the advertising their presence and the offered services through periodic broadcasts. The Wave Service Advertisements (WSA) control information sent by the RSUs over CCHs. The wave-based basic service set (WBSS) is setup after WSAs are sent data exchange over the SCHs can only occur after the vehicle successfully receives the WSA. The signal strength of the RSU should be tuned to the network latency and the lane speed at that location, so as to not miss any vehicles coming in its range.

It is expected that vehicle will send the GPS location using the DSRC Service Channel to relieve the RSU of too much computation and keep it real-time. For the GPS data from vehicle, channels are exclusively reserved, informed of this and is tuned to sufficient bandwidth. However, there are some concerns due to a separate channel dedicated like switching between channels can cause delay etc. which needs to be addressed, so the RSU will now be receiving the location updates from the vehicles. Using this information the RSU will determine the nearest vehicle to the intersection. There are two methods to compute the distance of the vehicle from the intersection as pair wise Computation method and pair wise computation using Historic data.

To avoid the shadowing effect sometimes when a larger building or a larger vehicle shadows other vehicles, we use the mechanism of piggybacking the WAVE based parameters which would further reduce the communication gap between the car and the RSUs.

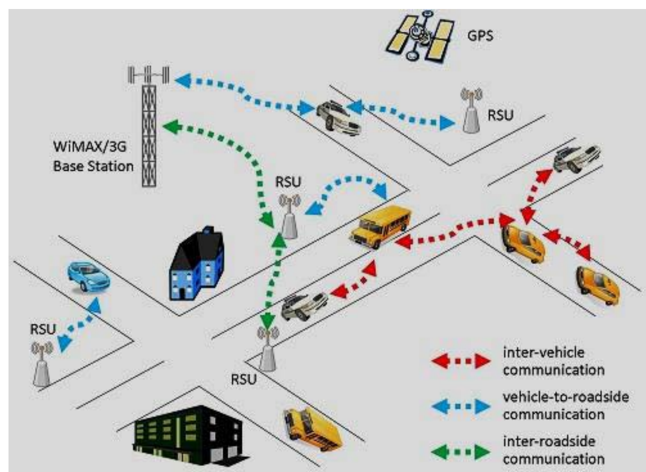

Figure 5: Car to Roadside Communication [23]

Rest of this paper is structured as Section-2 covers the related work. The proposed work in this has been discussed in section-3, numerical analysis and simulation results are discussed in section 4 and at last section 5 concludes this paper.

2.0 LOCATION AIDED ROUTING (LAR)

Generally, for VANET we use two types of routing protocols such as namely traditional topology and GPS position based routing protocols. Topology based routing protocol is divided into two category that is reactive and proactive routing protocols. Reactive routing protocols are more secure routing schemes for VANET and perform on-demand basis, in which source node initiates the route discovery process using flooding.

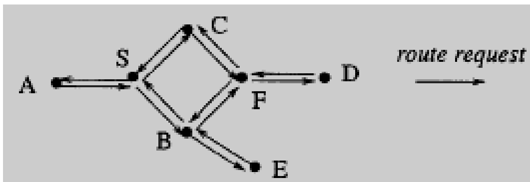

Figure 6: Message flooding

In figure 6 we can see when the source node S would like to communicate in the network with a destination node D. For this source node S will try to find out a route for destination node D in the network. For the same purpose source node Swill broadcast a message known as route request (RREQ) message to its all neighbors. After receiving the RREQ message, nodes Band C will forward the RREQ message to its all neighbors in the network. We can see in the figure 6 after receiving RREQ message from the node B, the node F will forward it to its all neighbors node. The node F will discard the RREQ message; however, it receives the same RREQ message from the node C. As RREQ message has been propagated to various nodes on the network, now path can be discovered on the basis of RREQ message. To decide that intended destination is reachable or not from the sender node flooding algorithm m can be used. The intended destination node will respond by sending a RREP message to the sender node after receiving RREQ message. The RREP message will always follow the same path that is obtained by route discovery process.

During some circumstances or some time it may be happened that intended destination node is not able to receive a RREQ message such that when it is not reachable from the sender node or it may be possible due to transmission errors in the network. In such cases route discovery process will be reinitiated in the network by the sender node. The sender node will set a timeout when it again initiates route discovery process. If sender node does not receive a RREP message during this timeout interval then a new route discovery process will be initiated by the sender node with the help of other sequence number that is different from the previous route discovery recalls. To detect the multiple receptions of the same route request, the sequence numbers play very important role in the route discovery process. Timeout occurs in both cases such that first one is if intended destination node does not receive a message of route request and second one is that if route reply message from the destination is lost during the route discovery process.

In vehicular ad hoc network designing of routing protocol is a crucial problem and several numbers of routing protocol algorithms have been developed to find out the suitable path in the network. Adaption of traffic patterns is a desirable qualitative property of the routing protocol [7].

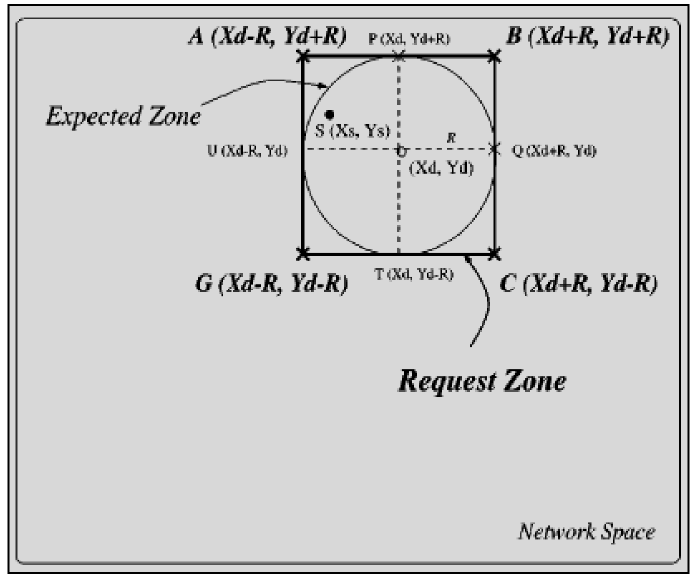

Figure 7: Source node outside the expected zone [20]

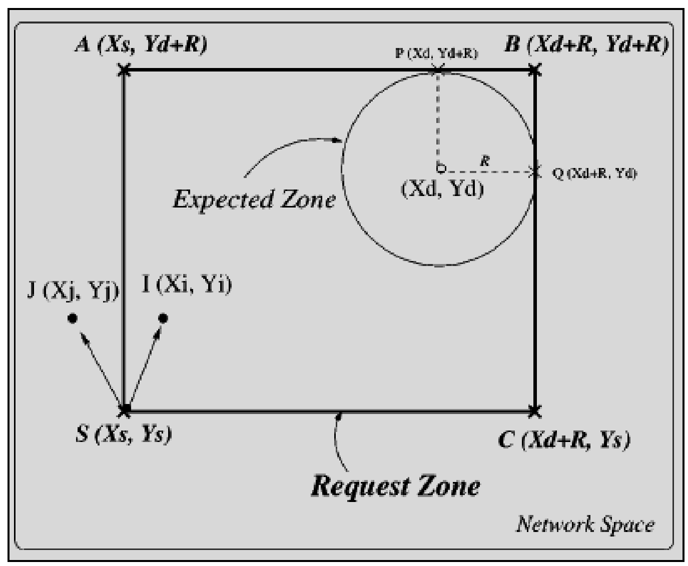

Figure 8: Source node inside the expected zone [20]

Maltz and Johnson [9, 11] has been discussed conventional routing protocols those are not sufficient for the ad-hoc networks. Protocols those use periodic updates of routing tables may waste the wireless bandwidth and amount of routing related traffic. They proposed a routing protocol based on the on-demand route discovery process by using Dynamic Source Routing (DSR).

If once request zone is formed in the LAR protocol, the neighboring nodes that are falling out of the request zone will be removed. Only neighboring nodes that exist in the request zone can accept and forward the request message for further processing in the network. Therefore, in order to control flooding and overhead in the network, LAR places a limit on the number of advancing nodes.

The routing protocol presented by Royer and Perkins in [12] is known as AODV routing protocol that uses a demand-driven route establishment procedure. The routing protocol designed by H. T , Karp, B. and Kung in [13] named TORA that minimize the reactions of topological changes in the network by localizing routing related messages to a small set of nodes near the change.

Hass and Pearlman in [14] have combined properties of the both routing protocol such as proactive and reactive approaches to design ZRP. The recent papers have presented comparative performance evaluation of several routing protocols. During the route discovery process in the vehicular ad hoc network Young-BaeKo and Nitin H. Vaidyain [17] are proposed a LAR routing protocol that uses concept of fractional flooding to decrease the control overhead. Position of the nodes in the vehicular ad hoc network is obtained by using a location finding system such as GPS. Determination of the next forwarding node in the network to establish a route, the Location Aided Routing protocol uses the position information of the node. Searching area in the LAR protocol for a node in the network to be restricted within a specific area, due to the limited route discovery area and condensed search area, route request (RREQ) packet is forwarded to an inadequate number of nodes gives reduced route discovery overhead.

Some of Location Aided Routing protocols have been proposed and these are needed geographical information for the selection of next forwarding nodes. Karp and Kung in [13]proposed routing protocol based on position of the nodes that uses geographical information of nodes in the network which are closest to intended destination node in order to forward the data packets and make successful communication. The measure drawback of this algorithm is that when it is evaluated for the large geographical area or large city, communication may be interrupted by the existing obstacles such as buildings in the city environments. Several other protocols have been defined in [14, 15] to overcome such problems. The proposed a position based routing protocols in [14] that use topology information to deliver the data from source to destination. In the next section, we have explained the minimization of one-hop delay that can be further used to enhance the performance of LAR protocol.

3.0. PROPOSED WORK

In this paper proposed model is known as LAR protocol that uses current location information of the nodes in the network. The basic purpose of this technique is to reduces routing overhead. Position of the node in the network can be obtained by using a device known as GPS [11]. By using such location finder device it is possible to know the physical location of mobile node. Position information of the node obtained by GPS always includes some errors which are the differences between the coordinates calculated by GPS and actual coordinates. The GPS NAVSTAR has positional accuracy of about 50 to100 meters and Differential GPS offers accuracies of a few meters [12]. In the proposed model, we have assumed that each node is aware with its current location precisely with no error.

In the proposed model each node knows its location and speed through some separate location service, such as GPS. Once when RREQ message is received at the destination node, the destination node unicast a RREP message back along with the reverse route found in the header. The LAR is a one of more popular routing protocol uses a concept known as LAR Box, in that a neighbor of source node, S, is determined if it is within the common transmission range of two intersecting circles in LAR box area. The radius of the transmission circle can be defined as radius, $R=V_{avg}\times (t_1-t_0)$ centered at (X_D, Y_D). In proposed model [16],author is discussed about the node distribution, velocity, link duration and path duration but distance related and average numbers of hop counts are not discussed.

Young-BaeKo and Nitin H. Vaidyain [17] suggested a substantial reduction in the number of routing messages by creating request zone and expected zone for LAR protocol. Basic objective of our proposed model is to find out average distance between source and next-hop nodes then we have calculated one-hop delay during the data transmission. If the destination node is already available in the source's transmission range then source node directly transfer the packet to destination using unicasting.

To forward information in the network when the destination is outside the sender's transmission range next-hop forwarding node (intermediate node) is required for further transmission [6].In order to reduce one-hop delay, we need to minimize the hop count from source node to next-hop node. In this case we can select the next-hop node which positioned at the maximum transmission range of sender node in the network.The figure 9 is showing the shaded area having the nodes positioned at the maximum distance (border) of the source node.

3.1. Probability of Next-Hop Node Selection

In figure 9 we can see that source node S would like to communicate on the network with the destination node D. The destination node D is out of the range of communication range of the source node S. In this situation to complete the communication intermediate next forwarder nodes are required. In our propose model we have considered vehicle to vehicle communication scenario that has no infrastructure or road side unit along the road side. To receive and transmit valuable information sensors are used on the vehicle. In the proposed model transmission range of every vehicle is assumed to be equal, denoted by R. In this communication link between two vehicles depend only on the distance between them means as long as vehicles are within the transmission range of each other. In VANETS, each vehicle is able to obtain won current location and velocity information because each vehicle to be equipped with GPS and digital road maps. Suppose there is K

nodes are arriving in the shaded area in figure 9 and they are following Poisson distribution process and can be calculated as follows:

$$P(K) = \sum_{N=K}^{\infty} \binom{N}{K} (p)^{K} (1-p)^{N-K} \left(\frac{(0.352 * \rho * X^2)^N * e^{-0.352*\rho*X^2}}{N!} \right)$$

Therefore,

$$P(K) = \frac{(0.352 * \rho * p * X^2)^N * e^{-0.352*\rho*p*X^2}}{N!} \quad (2)$$

We can find out the probability of at least *K* neighboring nodes consequently in the given area as shown in the above figure is:

$$P(K_a) = 1 - \sum_{i=0}^{K-1} \frac{(0.352 * \rho * p * X^2)^i * e^{-0.352*\rho*p*X^2}}{i!} \quad (3)$$

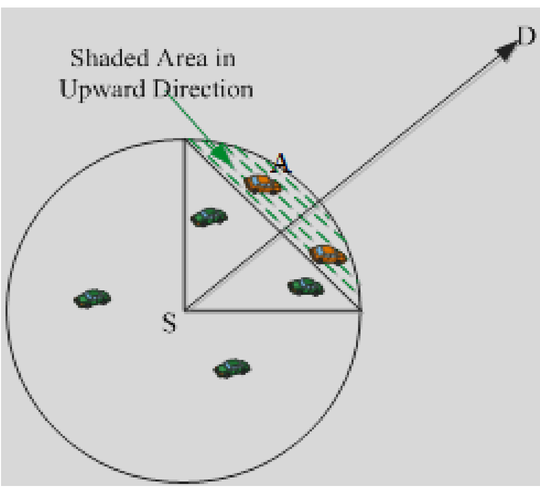

Figure 9: Selection of node with maximum distance

Now, we are able to calculate selection probability of at least one node in the shaded area within the transmission range *R* with the help of equation 3as following:

$$P = 1 - P(X = 0)$$

$$P = 1 - e^{-0.352*\rho*p*R^2} \quad (4)$$

3.2. Expected Distance Calculation

As shown in the figure9, we can see the first sender node is source node S and border node is node A. The border node A can be used as a next-hop forwarding node positioned at the maximum distance within transmission range R. Suppose there are n neighboring nodes of source node, S in the forward area towards the destination node, *D*. Let n-1 nodes out of n nodes are within shaded area and node is at maximum distance or closer to border of the sender's transmission range.Now node distributions of , i=1,2,…n. Suppose f(k) and F(k), where k=1,2,…n denotes the PDF and CDF of these distances. Thus probability distribution function (PDF) of finding the first neighbor can be given as:

$$F(k) = P[d1 \le k, d2 \le k, d3 \le k, \dots, d_n \le k]$$

$$F(k) = \prod_{i=1}^{n} (1)P[d_i \le k] = \left(\frac{k}{X}\right)^n = \frac{d}{dk} F(k)$$

$$F(k) = \frac{d}{dk}\left(\frac{k}{X}\right)^n = \frac{n}{R}\left(\frac{k}{X}\right)^{n-1} \quad (5)$$

Similarly, the expected value of *k* can be computed as:

$$E(k) = \int_0^X k.f(k)\,dk = \int_0^X k.\frac{n}{X}\left(\frac{k}{X}\right)^{n-1} dk$$

$$E(k) = \frac{n}{X^n}\left[\frac{x^{n+1}}{n+1}\right]_0^X = \frac{n}{X^n}\left[\frac{X^{n+1}}{n+1}\right] \quad (6)$$

3.3. Expected Delay

To improve the network performance in VANETs, it is required to select a suitable next-forwarder hop with suitable path in the network to forward data packets. To minimize the delay during data transmission, suitable routing protocol such as position based routing protocol communicates packets using radio waves as earliest. Since, in VANET, roads can be used as a medium for vehicular nodes through which the packet has to be transferred, therefore, the road with maximum velocity is selected first.

During data transmission, all the routing protocols in VANET assumes that smart vehicular nodes are furnished with computing device, sensors, digital maps, and advanced information processing tools. Digital map in the vehicle provide street and lane level map for drivers and traffic related information such as traffic density on the road, direction of nodes, position and velocity of vehicular nodes on the roads at disparate times of the day. Total delay is the time required to transmit data packet from source node to destination node. Therefore, the expected delay between two hops can be defined as:

1) $T_{delay} = Probabilityofatleastonenode * Speed + Probabilityc$

In the sender's transmission range, probability of at least one node can be given as:

$$P(x = 1) = (1 - e^{-\rho R}) \quad (7)$$

Similarly, the probability of no node in the transmission range is:

$$P(x = 0) = e^{-\rho R} \quad (8)$$

Therefore, the expected delay can be written as

1) $T_{delay} = (1 - e^{-\rho X}) \cdot \frac{E(k)}{S} + e^{-\rho X} \cdot \frac{E}{}$

Where,

E(k) = Expected Distance between two hops

X= Communication Range of the Node

S= Speed of vehicles

= Node density in the network

4.0. RESULT ANALYSIS

In this paper we have analyzed and presented the results for the above mathematical equations carried out with the help of MAT LAB simulation tool. In this results analysis section, first we have shown the distribution of nodes the given area for different values of node density. Further, we have computed the results in terms of expected delay with respect to communication range and speed of the node in the network.

4.1. Probability of Next Hop Selection

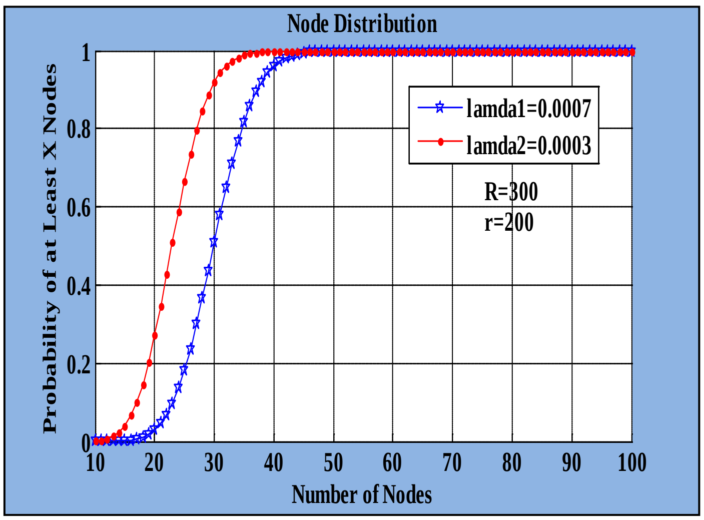

Figure 10: Next-hop selection probabilities

The figure10 is showing selection probability of the next- hop node with respect to number of nodes to forward data packets in the network. We can see in the figure, next hop selection probability is higher when range of nodes is lower and it's start decreasing when numbers of nodes are increasing in the network.

Also in figure11, we have shown next- hop node selection probability with respect to communication range; it is increasing as well as communication range is increasing.

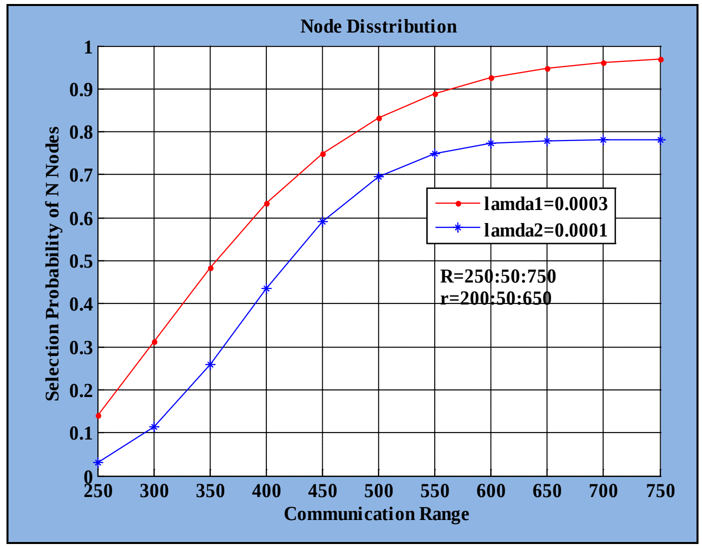

Figure 11: Next-hop selection probabilities

4.2. Expected One Hop Distance

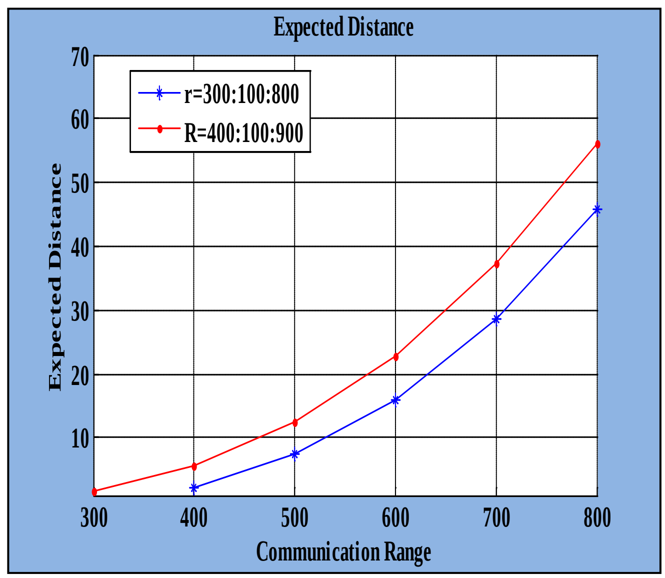

Figure 12: One hop expected distance with respect to communication range

The figure 12 is showing expected one hop distance between sender node and next forwarder node with respect to communication range. We can see in the figure expected one hop distance between sender and next forwarding node is increasing as well as communication range is increasing.

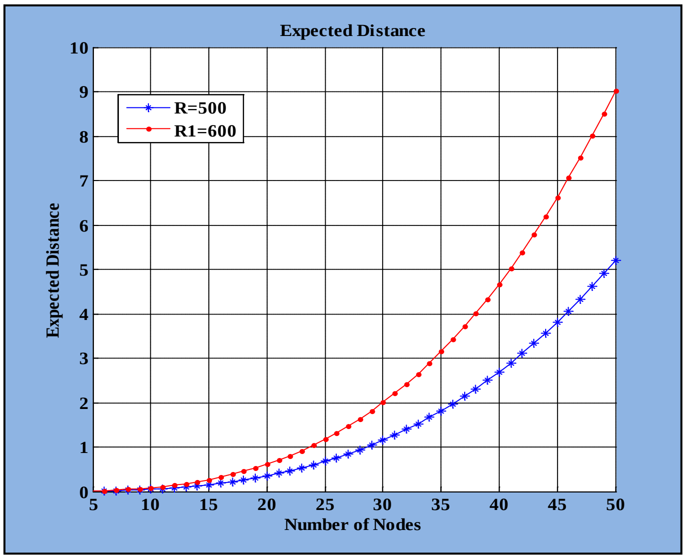

Figure 13: One hop expected distance with respect to nodes

The figure 13 shows expected one hop distance between sender node and next forwarder node with respect to number of nodes. We can see in the figure expected one hop distance between sender and next forwarding node is increasing as well as number of nodes are increasing.

4.3. Expected Delay

In figure 14, we have shown the graph for expected delay with respect to communication range. As shown in the figure, as the communication range increases, the expected delay between nodes decreases because of distance between the source and border node is also increasing.

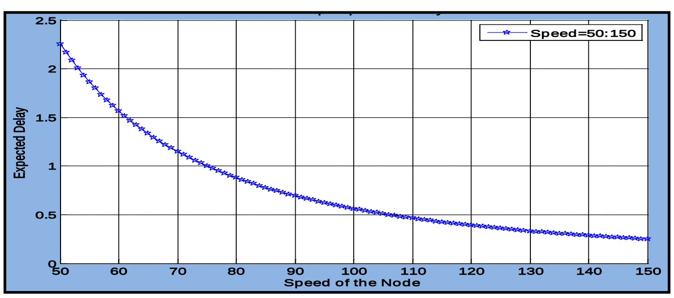

Figure 14: Expected delay vs. communication range

Further, we have computed the expected delay with respect to speed of the node in VANET. As, we know that, VANET is highly dynamic in nature. It means packet transmission is high due to high speed of the vehicular nodes in the networks. As shown in the figure 15, as the speed of the node is increasing, the expected delay is decreasing. It means packet delivery ration will be more.

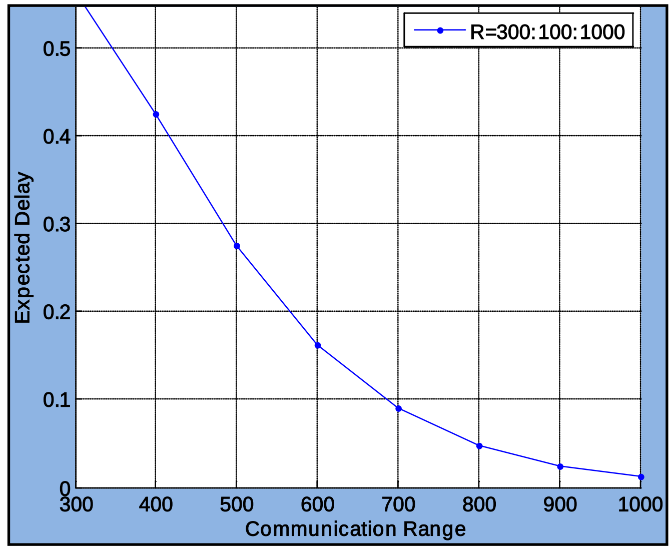

Figure 15: Expected delay vs. speed of node

5.0. CONCLUSION

In this work, we have discussed the position based routing protocols for VANET and used LAR protocol for mathematical calculations. We have analyzed LAR routing protocol by considering the border node as a next-hop forwarding node to transmit the packet from source to destination in the network. Further, we have computed the results in terms of expected delay with varying communication range and speed of nodes. The mathematical analysis and simulated results shows that the nominated next-hop neighboring node as a border node gives better performance in the network. Therefore, message can be forwarded appropriate to minimize the large number of road accidents and enhance the routing performance.

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Delay and Hop Count Estimation of Directional-Location Aided Routing Protocol for Vehicular Ad-hoc Networks

Kavita Pandey[1], Saurabh Kumar Raina[1] and **Ram Shringar Raw[2]**

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Abstract – Vehicular ad-hoc network (VANET) is an integral component of intelligent transportation systems. Due to continuous movement of vehicles, network formation and de-formation is very frequent. Furthermore, for safety applications, we wish to create the communication without any delay or collision. Therefore, communication among the vehicles is even more challenging. D-LAR (Directional–Location Aided Routing) is one of the position based routing protocol which works efficiently in VANETs. We want to analyze the performance of D-LAR protocol in a city traffic scenario. To do a quality performance evaluation, extensive simulations has been done in realistic environment created with SUMOatraffic simulator and NS-2 network simulator. The performance of D-LAR protocol has been compared with LAR on various metrics like routing overhead, packet delivery ratio and delay.

Index Terms - Delay, D-LAR, Hop count, LAR, Routing protocol, Vehicular ad-hoc networks

NOMENCLATURE

DIR - Directional Routing, D-LAR – Directional-Location Aided Routing, ITS- Intelligent Transportation Systems, LAR - Location Aided Routing, PDR – Packet Delivery Ratio, VANETs - Vehicular Ad-hoc Networks,

1.0 INTRODUCTION

Traffic and transport management systems are using the new generation information and communication technologies to have intelligent transportation systems (ITS). Their aim is to ease the citizen's Life by providing them facilities like smart parking, intelligent traffic management systems,etc.VANETacts as a vital innovationfor ITS. It's a kindof network where vehicles can communicate with eachother as per therequirements.

To do this, intelligentvehicles that have wireless transceivers and computerized control modules are required.

There are two kinds of communication in VANETs. One is vehicle-to-vehicle communication which is among vehicles only. Another type of communication is vehicle-to-infrastructure which is among the vehicles and roadsideinfrastructure units.

Here,we are considering only vehicle-to-vehiclecommunication which is purely ad-hoc and completely infrastructure-less. The two vehicles can communicate when they lie in the communication range of each other. Vehicles movements are restricted as per the road-maps. Another important characteristic of VANETs is uneven distributions of vehicles on the roads. On top of that vehicles speed is irregular and generally it is high, therefore network formation and de-formation is very frequent. Due to all these characteristics, doing the communication and maintaining it is very challenging.

A number of researchers introduced a variety of routing protocols. These protocols have been grouped into two categories: position based and topology based routing protocols. Due to various characteristics of VANETs mentioned in the above paragraph, topology based routing protocols such as DSR, DSDV and AODV have poor adequacy and adaptability.

Position based routing protocols works on the concept of greedy forwarding. Here, each node knows the positional coordinates of its neighboring nodes. After getting the positional information, source node selects the one as a next hop node who is geographically nearest to the destination. These protocols are also called as geographic or spatial aware routing protocols. In position based routing protocols, there is no requirement of establishment and maintenance of route. Also, these protocols do not require the knowledge about the complete network connectivity. These protocols utilize the spatial knowledge like lanes and maps of cities for routing decisions. Therefore, position based routing protocols performance is better than topology based protocols.

However, position based routing protocols require extra information in terms of its own position, neighboring nodes and destination position. To get its own position, all the nodes utilize the Global Positioning System. To get the destination node positional information, source node takes the help of any location service. After getting the destination position, source node mention it in the packet's destination address. Neighboring nodes gather the positional information of each other by periodic exchange of beacon or hello messages. GPSR[HYPERLINK \l "BKa00" *1*], A-STAR*2*]**}**, GEDIR[HYPERLINK \l "Iva99" *3*], LAR*4*]**}**, GSR[HYPERLINK \l "CLo05" *5*], D-LAR *6*]**}**etc. are different examples of position based routing protocols.

D-LAR (Directional-Location Aided Routing) [HYPERLINK \l "Ram12" *6*]proposed by Raw etal.is a position based routing protocol. This is amalgamation of two routing protocols, LAR and DIR. LAR (Location Aided Routing) protocol has been

[1]Department of Computer Science and Information Technology, Jaypee Institute of Information Technology, Noida, U.P., India.
Email: kavita.pandey@jiit.ac.in,rainsak@gmail.com
[2]AmbedkarInstitute of Advanced Communication Technologies and Research, Delhi, India.
Email: rsrao08@yahoo.in

proposed by Young and Vaidya*4*]**}**. In this protocol, they proposed the ideaof limited flooding for reducing the route discovery overhead. By getting the destination location and speed information, source node constructs an area, called as request zone which covers both source and destination. Now, route discovery is restricted to this constructed request zone. It helps in reducing the route discovery overhead. D-LAR protocol utilizes this concept for limiting the search area of route selection. Once the search area has been limited, now next-hop node would be selected from this search area. For the selection of next-hop node,D-LAR uses the concept of directional routing (DIR). D-LAR selects the next-hop node amongst the available neighboring nodes that has the minimum angular deviation from the connecting line of source and destination.

Raw et al. [HYPERLINK \l "Raw15" *7*]proposed D-LAR with a mental make-up that it achieves a remarkable performance in city traffic scenario because routing overhead would be reduced with the partial flooding concept of LAR protocol and directional greedy forwarding provides a stable route in comparison to other greedy forwarding techniques. They have provided the results of D-LAR for hop count and path throughput metrics. As we know that delay is an important metric for safety applications of VANETsbut they have not validated their proposed idea for this metric, while proposing the idea they said it is better for safety applications. D-LAR protocol is based on LAR even then D-LAR results have not been compared with the LAR protocol.

In this work, we have evaluated the performance of D-LAR protocol in terms of delay, packet delivery ratio and routing overhead metrics. Its performance has beencompared with the existing LAR protocol. We have also analyzed how delay gets affected by the both MAC layer and routing layer parameters.

The remaining article is organized as follows. The next section presents a brief overview of D-LAR routing protocol. Section 3 presents the review of some articles where authors did the mathematical modeling of delay with respect to wireless networks. Hop Count and delay analysis is provided in section 4. Section 5 and 6 presents the simulation results and conclusion respectively.

2.0 OVERVIEW OF D-LAR PROTOCOL

Among the available position based routing protocols in the literature *8*]**}**[HYPERLINK \l "Ber09" *9*]**,** D-LAR is one of the most suitable position based routing protocol specially meant for highly dynamic dense networks. D-LAR is the extension of LAR protocol. LAR limits the route discovery area with the knowledge of speed and location of destination. Suppose, the destination node **}**is moving with the speed and at time , it is at location. With this information source node can expect that the destination node would be somewhere in the circular region around attime . This circular region is of radius . This region is called as expected zone where source node can expect that destination node would be at time . After constructing the expected zone, source node startsdefining the request zone. Request zone is the smallest rectangle whose bottom left corner is the current location of and top right corner covers the expected zone. The sides of the request zone are parallel to X and Y axis.

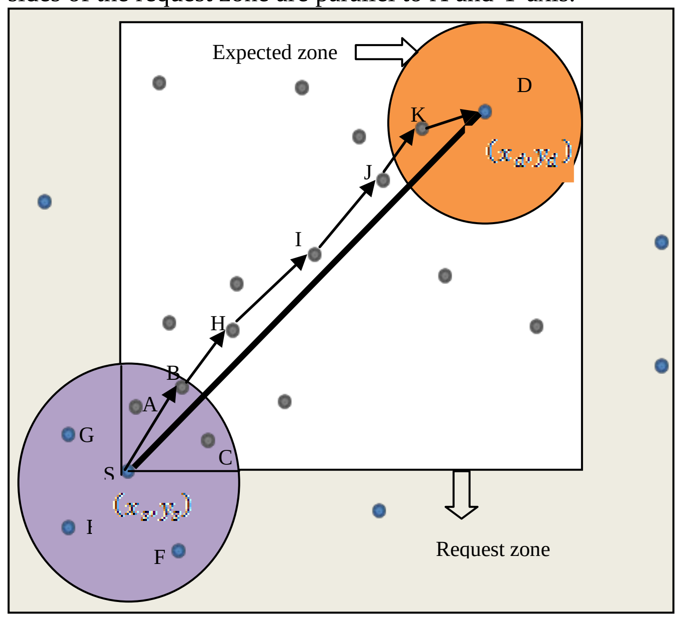

Figure 1: Working of D-LAR protocol

Source node initiates the route discovery process by broadcasting a route request message to its neighbors. Route request message encompasses the coordinates of the request zone. When a neighboring node receives the route request message, it checks its own positional information with the content of the route request message. If the neighboring node lies within the request zone it forwards the message further to its neighbors else it will drop the message.

It can be visualized from figure 1 that A, B, C, G, I and F are the neighbors of S. Source node sends the route request message to all its neighbors. However, only three nodes A, B and C propagate the message further because they lie in the request zone. In this way the message is not propagated to all the nodes of the network. There is an involvement of few nodes which lies in the request zone. The route would be searched in the direction of destination; as a result the routing overhead is reduced. S would choose node B as the next-hop node because it has the minimum angular deviation in comparison to other two neighboring nodes. By using the same selection criteria, node B will select the next-hop node as H and then similarly H selects I and so on.

This concept of LAR is used by D-LAR protocol for broadcasting of control packets, so that within the request zone all the nodes knew the location information about itself and its neighbors. After this, D-LAR selects the next-hop node among the neighboring nodes within the transmission range and the request zone. The next-hop node would be the one which has the minimum angular deviation from the line connecting source to destination. By using this next-hop selection strategy, message will not go out of way.

By using the strategy of LAR, D-LAR routing overhead is reduced and with the use of directional greedy forwarding, the selected path is always stable. Both routing overhead and stable path are desirable parameters for safety applications of VANETs.In all applications of VANETs but specifically in safety applications, delay is very important measuring metric for evaluation of the performance of a routing protocol. If the information is delivered in a timely manner then the driver got the time to react in order to avoid the problem ahead.
Raw et al. asserted the performance of D-LAR protocol in city environment although they have not provided any proof for their claim in terms of delay, the important performance measuring metric. It has not been tested in a real traffic simulation. Therefore, the mathematical analysis of delay for D-LAR protocol has been done. The performance of D-LAR protocol has been tested in a real traffic scenario.

3.0 RELATED WORK

This section presents an overview of articles where researchers analyzed the delay with respect to different conditions. In the first article, Abdullah et al. [*10*] has shown the impact of hidden nodes on delay and throughput metrics. They have assumed that interference range, transmission range and sensing range, all are equal. Therefore, the communication area is divided into two parts: hidden node area and other area. As per these considerations, they have calculated node transmission and collision probabilities.
In another article, Ding etal. [*11*]presenteda model for a wireless node using Markov chain analysis and derived the various probabilities like idle, successful transmission etc. as per state transitions in respect of multi-hop ad-hoc networks. They have measured the delay by calculating the steady state probabilities of each state and time duration spent by each state. Bianchi etal.[*12*] have analyzed the performance of 802.11 DCF in terms of throughput for both basic access and RTS/CTS mechanisms. They have assumed the finite number of nodes and ideal channel conditions for the calculation. Also, they have calculated the various probabilities and throughput performance of the system with saturated nodes by providing the Markov chain model for back-off window size. In another work, a Markov chain model for single node using IEEE 802.11 has been presented by Ghadimiet. al. [*13*]. This model is established on the Bianchi's work [*12*]. Here they measured end-to-end delay for multi-hop wireless ad hoc networks by considering hidden and exposed terminals conditions under unsaturated traffic.
In all the above cited articles researchers have figured out either the delay or throughput concerning various traffic situations and channel conditions. Performance of any system can be estimated by considering the node transition probabilities like collision, transmission, idle, etc. The Markov chain model is the prominent model to measure the performance of any system. In this article, we have approached in the same way by considering the real traffic scenario for VANETs and all the simulations has been done using SUMO[*14*] as a traffic simulator.

4.0 HOP COUNT AND DELAY ESTIMATION

Path delay is also labeled as end-to-end delay; the delay a packet experiences before its successful delivery at destination. If the expected delay with respect to one link is known then path delay can be computed as,

$$d_p = H * d_H \qquad (1)$$

Where*H* is the expected number of hops from source to destination that can be calculated from the Hop Count Algorithm described in the sub-section 4.1.   is one hop delay that is further examined in sub-section 4.2.

4.1 Hop Count Estimation

For the calculation of H, refer the algorithm proposed by pandey et al. [*15*] [*16*]. Here, that algorithm has been used with one difference that is estimation of distance covered with in one hop. Therefore, the expected distance is calculated here. Suppose, expected $\mathcal{D}_{si}$(estimated one hop distance)is represented by a random variable *x*.
There is an assumption that ‘ ’neighboring nodes are available in the forwarding region and transmission range of *S*, the source node. $\mathcal{D}_{si}$is a random variable that represents the Euclidean distance between the current node *s* (first time it is source, next time onwards the chosen next-hop node in previous step) and the chosen next-hop node *i*(according to the criteria of minimum angular deviation). Therefore, cumulative distribution function of $\mathcal{D}_{si}$can be written as [*17*]

$$F_{\mathcal{D}_{si}}(x) = P(\mathcal{D}_{si} \le x) = \frac{\pi x^2}{\pi R^2} = \frac{x^2}{R^2} \qquad (2)$$

Thus, the probability distribution function of $\mathcal{D}_{si}$is

$$f_{\mathcal{D}_{si}}(x) = \frac{dF_{\mathcal{D}_{si}}(x)}{dx} = \frac{2x}{R^2} \qquad (3)$$

Expected one hop distance, $\mathcal{D}_{si}$can be calculated as

$$E(\mathcal{D}_{si}) = \int_0^R x\frac{2x}{R^2}dx = \left|\frac{2x^3}{3R^2}\right|_0^R = \frac{2R}{3} \qquad (4)$$

Hop count value computed with the help of algorithm and expected one hop distance would be used in the following subsection for calculation of delay.

4.2 Delay Estimation

One hop delay can be dictated as the summation of all types of delays the packet experiences while travelling from one node to the next-hop node. It can be written as

$$d_H = Trans.delay\ (d_T) + Prop.delay\ (d_{Pr}) + Queuing\ delay\ +\ Pro$$

Here, a city traffic scenario has been considered where average speed of vehicles is approximately 50 km/hr. Therefore, propagation delay can be written as

$$Prop\ delay\ (d_{Pr}) = \frac{one\ hop\ distance}{avg\ speed} = \frac{\mathcal{D}_{si}}{50}.$$

Therefore, from the equation (4),   can be written as

$$d_{Pr} = \frac{R}{75} \quad (6)$$

Processing delay is assumed to be negligible. Queuing delay is the period of time that a packet experiences at the interface queue. At the most a node can have one packet waiting for transmission so queuing delay is ignored.

Transmission delay is defined as the time when the packet is ready for communication until it is transmitted successfully. Successful transmission is possible only if at an instant of time only one node is transmitting and the channel is error-free. If two or more nodes will transmit at the same time then collision will occur. Sometimes, transmission gets interfered due to hidden nodes or interfering nodes. In both the cases, there is a need of retransmission of packet. So, transmission delay accounts for the wasted time due to failed transmission and time taken in successful transmission. It can be written as

$$d_T = T_C + T_S.$$

is the wasted time for a failed transmission due to collisions and is the duration of successful transmission. Therefore, one hop delay can be re-written as

$$d_H = T_C + T_S + d_{Pr}\ .$$

5.0 RESULTS AND PERFORMANCE ANALYSIS

NS-2, network simulator and SUMO, a traffic simulator has been used to validate the proposed concept. Realistic city traffic environment scenarios have been designed using SUMO. All the vehicles were deployed as per the road constraints in an area of 600 * 800 m^2. The whole area has 20 junctions. Figure 2 shows a small segment of the map and vehicle movements on the roads. The yellow objects are depicting as vehicles. Different scenarios have been created by varying the number of vehicles.

Simulation results have been presented in the form of three metrics by varying the number of vehicles. Results of D-LAR protocol have been compared with LAR protocol on routing overhead, delay and packet delivery ratio metrics. In the graphs, each result value is an average of 5 iterations so the complete simulation has been run 5 times, therefore 5 different route and mobility files have beengenerated using various python files of SUMO. These files were converted into NS-2 [*18*] supported mobility files using traceExporter.py, python file of SUMO.

All the parameters used in our simulations are summarized in table 1.Routing overhead is an important metric because of extensive change in topology in VANETs. It is defined as the number of control packets to be sent to receive the sent data packets.

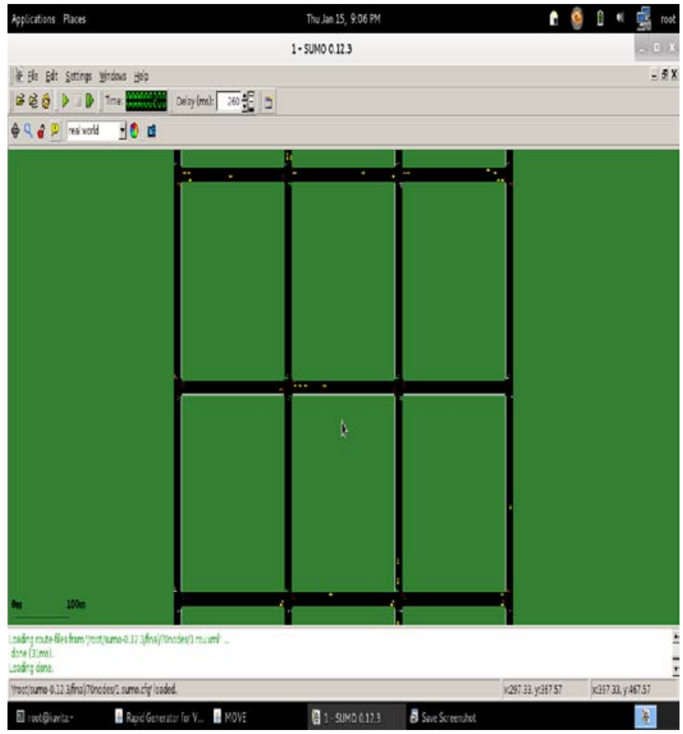

Figure 2: Map and vehicle movements in SUMO

Parameters	Value
Network Simulator	NS-2.32
Traffic Simulator	SUMO-0.21
Simulation area	600 * 800 m^2
Simulation time	1000 sec / iteration
Propagation Model	Two-ray ground
Transmission range	250 meters
Traffic source	CBR
MAC protocol	IEEE 802.11 DCF
Routing Protocol	D-LAR, LAR
Transport Protocol	UDP
Data Packet Size	512 Bytes
Packet rate	4 packets/sec
Bandwidth	2 Mbps
Number of Vehicles	100 to 300 with a variation step 50
Speed	50 km/hr.

Table 1: Simulations parameters

As we have already explained that delay is an important metric especially for safety applications in VANETs. Therefore, we have chosen this metric for validation of proposed concept. Delay value shows the average amount of time taken by the packets to reach from source to destination. The performances of both the protocols, D-LAR and LAR have also been compared on packet delivery ratio. The ratio of successfully received data packets upon sent data packets is called as packet delivery ratio (PDR).

All the three figures 3, 4 and 5show the better performance of D-LAR over LAR protocol. Figure 3 shows the routing overhead of D-LAR is less in comparison to LAR. It is because of the routing strategy of D-LARi.e. a node which is present in the request zone and having minimum angle will forward the packet. Whereas in LAR, all nodes present in the request zone forward the packet further. As the number of nodes increases, the probability of more number of nodes in the request zone also increases and therefore as a result routing overhead increases.

Figure 4 presents the average end-to-end delay results against number of nodes. It can be visualized that delay of D-LAR protocol is better than LAR and its value increases as the node density increases. This is due to the fact that with an increase in node density nodes in the request zone area increases and therefore collision probability increases. As a result, delay value increases.

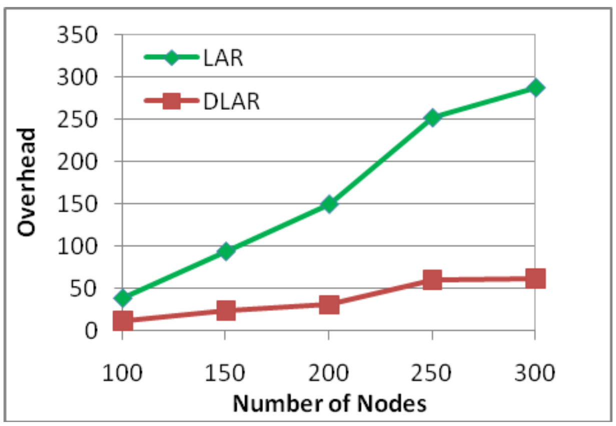

Figure 3: Routing overhead vs. number of nodes

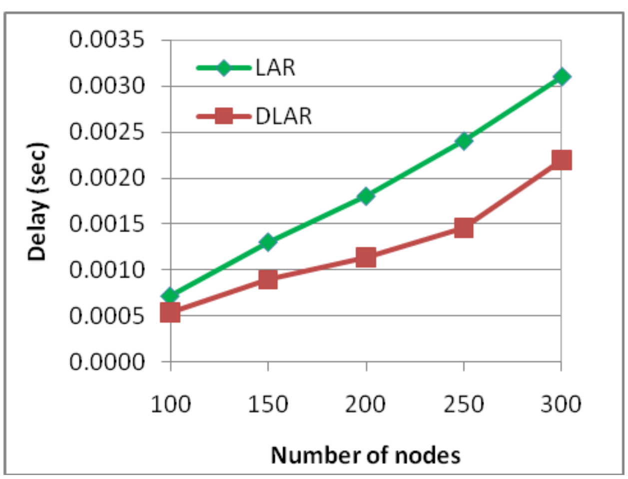

Figure 4: Delay vs. number of nodes

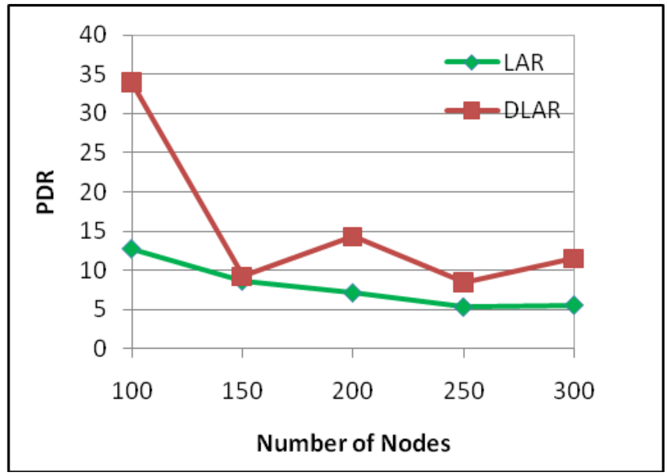

Figure 5: PDR vs. number of nodes

Figure 5 demonstratesthe PDR results against the node density. The packet delivery ratio of both the protocols decreases with an increase in number of nodes. It is because as the node density increases, the probability of collision among the data packets also increases, which results into a lower packet delivery ratio. Even though, PDR of D-LAR is better than LAR. From all the results, it can be said that D-LAR performance is better than LAR.

6.0 CONCLUSION

In this article, we have mainly analyzed the hop count and delay for D-LAR position based routing protocol. D-LAR is a mixture of DIR greedy forwarding and LAR protocol. So, D-LAR protocol performance has been compared with the LAR protocol on routing overhead, packet delivery ratio and delay metrics. For a quality performance evaluation, simulations have been done using SUMO and NS-2. Various scenarios have been created by differing the number of vehicles. It can be visualized from the results that with an increase in number of vehicles, collision probability increases, so delay increases and PDR decreases. As per the routing strategy of D-LAR, its routing overhead is less in comparison to LAR. Form the results it has been verified that D-LAR is better than LAR, so, it is suitable for city traffic environment for VANETs.

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Barriers to Cloud Computing Adoption for SMEs in Saudi Arabia

Abdullah Basahel[1], **Mohammad Yamin**[1] and **Abdullah Drijan**[2]

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Abstract – The purpose of this article is to study the business model of cloud computing and the factors affecting its adoption in Small and Medium Enterprises(SMEs) in the west coast of Saudi Arabia (KSA). In this paper, our focus is to investigate find an answer to the question as to what is stopping SME's in Saudi Arabia from adopting Cloud Computing given the substantial financial and technical benefits it offers.

To provide comprehensive information, we have conducted a thorough analysis of Cloud Computing adoption in the global market and then compared it with that of the Middle East and North Africa. This comparison is carried out by utilisingthe market share and total spending methods. The data of the current state of Cloud adoption in KSA was dissected and profiled in order to reach meaningful conclusions. In this study;we were able to compare the economic benefits that Cloud Computing has brought to the KSA and the future benefits, which are yet to be realized. Based on an extensive survey of 45 small and medium enterprises in Jeddah, we find that still are some concerns of security and privacy, and there is a general lack of awareness about the benefits. These hurdles are hampering the proliferation of Cloud Computing into SMEs. Throughout this research, we have followed the descriptive method that best suits the nature of the research and the discectomy of the survey. Upon receiving the survey answers, a cross-section analysis was conducted across various industries to understand how the barriers to Cloud adoption vary from one setting to another.

Index Terms – Cloud Computing Adoption, SMEs, Middle East, North Africa, West coast of Saudi Arabia

1.0 INTRODUCTION

Cloud Computing is "an on-demand suite of infrastructure, server, storage, applications and information, which anybody, individuals, businesses or corporations can rent for a fee". Cloud computing is becoming a game changer for Small-Medium Enterprises (SMEs) by offering scalable and affordable infrastructure and capabilities available as services.

[1]Department of MIS, Faculty of Economics and Administration, King Abdulaziz University, Jeddah, Saudi Arabia
Abasahl@kau.edu.sa, myamin@kau.edu.sa
[2]Department of MIS, Faculty of Economics and Administration, King Abdulaziz University, Jeddah, Saudi Arabia
aldrijan@gmail.com

Hence, utilizing cloud technologies seems to be one the most attractive methodologies of cost minimization in SME's. Microsoft conducted a recent study of 3,000 SME's across 16 countries with the objective of understanding whether SME's has an appetite for cloud computing concluded that "43% of workloads will become paid Cloud services" [1]. The rapid Small and Medium Enterprises can now tap into virtually unlimited computing resources only when needed, thus, significantly reducing the costs of building and sustaining the systems infrastructure. Cloud models are delivering on the promise of helping businesses work smarter by providing flexible, cost-effective access to technology and information. adoption of cloud computing as forecasted by Microsoft studies is not very far from the truth. According to [2], spending on cloud computing in enterprises is five times the rates of spending on other traditional IT systems. However, he argues that the cloud computing adoptions in countries' rates are directly related to the technological development level in each. In fact, the literature on cloud computing has been vastly concentrated on developed countries, both regarding theoretical frameworks to understanding the adoption of cloud computing in SME's and regarding empirical studies that quantifies this impact[2]. This research paper is an effort to fill the gap by empirically investigating the cloud computing adoption in Saudi Arabia through a set of surveys and interviews of SMEs in KSA.

According to [2], more than 70% of the respondents in the telecommunication & information technology sector answered yes when asked about the migration plan of their organization services and data to cloud computing. Similarly, 46% of the respondents in the manufacturing sector also answered yes. These results are positive indicators that the Saudi enterprises are not very far from embracing cloud computing and changing their operations to the cloud. These results have served as an effective starting point for this study.

1.1 Research Question

In this article, the main factors affecting the adopting of a cloud-computing in SMEs in the west coast of the KSA which includes the commercial and post city Jeddah, is going to be investigated. The KSA government has been actively supporting entrepreneurs by forms of subsidies, competitions and various other mentorship and acceleration programs. Through understanding the dynamics of the decision-making process related to cloud computing, this research can complement our efforts to create more value within the country. Ultimately, the objective of this study is to help start-ups, entrepreneurs and SMEs to reap the cost-effective benefits of cloud computing that would boost their performance, increase their profits and allow them to compete globally. The central question that this research is to investigate is as to *what is stopping SME's in Saudi Arabia from adopting cloud computing given the substantial financial and technical benefits it can add*. The main target audiences for this research are the entrepreneurs, owners and founders of small businesses, decision makers in medium sized companies, and government officials that subsidies any form of cloud computing model.

Many would argue that SMEs are the biggest winners in utilizing cloud computing to power up their operations. By definition, SME's are focused on cost minimizing and profit maximizing. A study conducted on 1,242 IT professionals by CDW found that "88% of cloud users pointed to cost savings and 56% of respondents agreed that cloud services have helped them boost profits" [3]. Additionally, 60% of respondents said cloud computing has reduced the need for their IT team to maintain infrastructure, giving them more time to focus on strategy and innovation. Moreover, "62% of the companies that have saved money are reinvesting those savings back into the business to increase headcount, boost wages and drive product innovation" [3]. Comparing the theoretical benefits of cloud computing to SME's can almost always lead to one conclusion - the adoption rates are likely to be high. However, as we will investigate in later sections, SME's in Saudi Arabia are still hesitant to adopt cloud-computing technologies. The focus of this research is not to examine whether SMEs are willing to adopt cloud computing as a viable resource and business strategy; but rather why are they willing to adopt it or not. This research seeks to pose these questions on a representative sample of small enterprises in Jeddah, Saudi Arabia to examine the reasons behind the resistance to adopting cloud computing regardless of the substantial potential value it may add to their organizations.

2.0 CLOUDTECHNOLOGY

A lot of literature is available on the definition and models of Cloud Computing [4] – [6].Different aspects of cloud computing in Saudi Arabia have been studied by a number of researchers, which can be found in [2] and [7]. In summary, cloud technology is now matured and is being used extensively. This technology not only provides storage at affordable rates, it also provides infrastructure and services. Clouds are classified as public, private and hybrid. By resorting to cloud computing, one can start a new business with a relatively small capital. This is a huge benefit when we compare similar endeavours a couple of decades ago. Although much of the fears and apprehensions have been addressed adequately and effectively but in some parts of the world they still exist. Gartner [8] every year publishes hyper cycle of leading technologies. It's the hybrid cloud which is making a headline in Gartner reports of 2015 [8]. As in [4], the cloud computing technology has already attained its peak. However, many developing countries are still lagging behind and missing on benefits of this smart technology.

2.1 Cloud Growth

With the growth and various models for the cloud, it is very beneficial to understand how these growth rates are structuredin relation to the cloud models. It is predicted that by 2018, 59% of the total cloud usage will be Software-as-a-Service (SaaS) as compared with 41% in 2013 [9]. On the other hand, Cisco [10] is predicting that 28% of the total cloud usage will be Infrastructure-as-a-Service (IaaS) in 2018 as compared to to 44% in 2013. Finally, 13% of the total cloud usage will be Platform-as-a-Service (PaaS) as compared to 15% in 2013. Figure 3below summarizes the growth patterns of SaaS, PaaS and IaaS over the period starting from 2013 till 2018 (Cisco, 2015) as in Figure 1

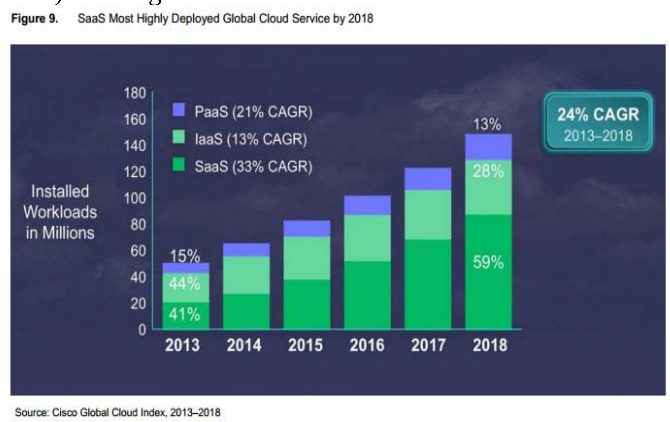

Figure 1: Cloud Growth

Steering away from ratios and percentage growth, the actual amounts of spending on cloud computing technology can be of great importance as it facilitates a more realistic comparison between the global cloud computing adoption as opposed to the regional and finally local amounts spent. Most of the research available recently is focusing more on the SaaS model, and since SaaS is the winner in terms of growth projections, it would serve as validbase of comparison between the global, regional and local market. According to Centaur Partners' analysis [8] of SaaS and cloud-based business application financial sheets and projections, the SaaS global market has grown from $13.5B in 2011 to $32.8B in 2016, achieving a 19.5% CAGR. In Centaur Partners' research, certain categories were picked to represent the SaaS industry including : 1) Content Management 2) Communication and Collaboration 3) CRM and ERP 4) Office Suites and Project Management Suites. Figure 4 below details the growth of SaaS-based businesses' revenues vs. the worldwide SaaS and Cloud Software revenues growth for the period of 2011 to 2017.

2.2 Clouds and ERP

ERP is a class of software that can help business owners run all business operations on one system. ERP software can handle Financial Management, Project Management, Warehouse Management, Supply Chain Management, Human Resource Management, Reporting Management, Document Management, Email Integration and Customer Relationship Management. All these functions are designed and integrated to work with each other. To add a function to an ERP solution, you just plug it and set it up in minutes. Cloud technology and ERP are benefiting from each other.

3.0 METHOD

We conducted a survey of a number of businesses in Jeddah, the commercial centre and the main port city of Saudi Arabia. A major challenge in getting data was the fact that micro enterprises are very conservative in nature. Therefore, a neutral, non-threating survey had to be distributed through proper channel which was familiar to the respondents. Another challenge emerged through the interviews for the first subject related to their company turnover- in order to profile the companies. An approach was the formulated appropriately to gather unbiased data. A thorough analysis of the previous surveys posted by large organization bodies including the Euro Commission were studied and analysed. Pre-survey interviews were then conducted to test the validity of questions and rule out any bias that could have occurred.

3.1 Survey Content

Understanding the rationale behind selecting the questions is crucial to understanding and analysing the results. In this section, each question will be briefly presented and detailed. Table 1 illustrates the ten survey questions presented to respondents.

Table 1: Survey Questions

Q1. What industry is your business part of?	Q2. Which of the following best describes your current job level?
Q3. What was the approximate revenue of your company in the past financial year?	Q4. What proportions of your total costs relate to Information Technology?
Q5. How familiar would you say are with what the cloud computing is?	Q6. Do you consider security and privacy concerns to be main barrier for cloud technology deployment
Q7. Which of the following do you perceive as main barrier to you using Cloud Services?	QUESTION 8: what would be the main driver for you to deploy cloud technology?
Q9. Which of the following service types you would consider to deploy?	Q10. Do you plan to invest in cloud computing in short, medium or long term?

3.2 Survey Results

The first question was a proxy question to understand the nature of industries where the start-ups come from. We found that more than 20% of the respondents were from the Information Technology background. This was followed by the Education sector, then Manufacturing and finally Communications fields and services. The responses to the following question were filtered through these all these four fields:

Which of the following would you perceive as a main barrier to using cloud services? The result are summarised in Tables 2-5.

Table 2: Information Technology

Answer Choices	*Percentage*	*Responses*
Security	72.73%	8
Service Availability Concerns	36.36%	4
Initial Investments/ Capital	36.36%	4
Provider Lock-in	9.09%	1
Loss of Control	36.36%	4
Training and Recruiting People to Deploy, Run and Maintain the service	27.27%	3

Table 3: Education

Answer Choices	Percentage	Responses
Security	100.00%	5
Service Availability Concerns	40.00%	2
Initial Investments/ Capital	40.00%	2
Provider Lock-in	0.00%	0
Loss of Control	20.00%	1
Training and Recruiting People to Deploy, Run and Maintain the service	40.00%	2

Table 4: Manufacturing

Answer Choices	Percentage	Responses
Security	57.14%	4
Service Availability Concerns	28.57%	2
Initial Investments/ Capital	42.86%	3
Provider Lock-in	14.29%	1
Loss of Control	28.57%	2
Training and Recruiting People to Deploy, Run and Maintain the service	42.86%	3

Table 5: Communications

Answer Choices	Percentage	Responses
Security	80.00%	4
Service Availability Concerns	40.00%	2
Initial Investments/ Capital	20.00%	1
Provider Lock-in	0	0
Loss of Control	60.00%	3
Training and Recruiting People to	40.00%	2

Deploy, Run and Maintain the service

4.0 RESULTS

From the results of the surveys, it is evident that most of the micro businesses in KSA believe that security is a major concern when it comes to adopting the cloud technologies. It is the same with micro businesses that spend an average 30% average of their spending on IT and Technology and are somehow familiar with the cloud computing model. With the findings various industries answered that Initial Investments, Cost of Hiring and Training and Service Availability are also constraining the proliferation of cloud technologies in the kingdom. Therefore, it can be inferred that the sloppy growth of cloud adoption in major parts of the KSA does have distinct reasons, with the security and privacy concern being the most prominent of them.

With the dramatic benefits, cost minimization and profit maximization opportunities that the cloud brings to micro-businesses, a lot of attention from the authorities is required to realize these benefits in the growing sector of micro-businesses. Specifically, the policy decisions undertaken by the government can boost and reshape the cloud market in the country.

4.1 Limitations

This is a quantitative survey of only 45 companies in a limited geographical area. Nonetheless this data provides important insights that can be generalized and highlighted. There is undoubtedly still a need for further investigations and rigorous studies and analysis to be able to come up with actionable data that can be passed to policy makers. Further research would be required to devise policies and create initiatives that would impact the cloud adoption in KSA.

4.2 Conclusion and Recommendations

The cloud computing business model is a viable business model with various added-values to the individual, the company and the country. From our analysis of the survey responses, we can say that the major barriers behind adopting cloud computing in the KSA tend to security and privacy concerns - micro businesses in KSA are still hesitant to share their data with cloud service providers, despite the well-established facts that the cloud is generally more secure than in-house computing. Secondly, in the telecommunication industry, service availability is a major barrier to the adoption of cloud computing. Thirdly, initial investments - with most of the respondents spending more than thirty-percent of their budget on IT, they lack appropriate funding for cloud computing – although it may be considered as a mismanagement of resources. Fourthly, recruitment of skilled personnel to run and maintain the infrastructure is critical to the success. Policy makers should systematically tackle these issues and empower the SMEs to benefit from the technology. Yamin [2] believes that the cloud computing model is administratively and dynamically similar to the outsourcing model As a way forward to the identified challenges, the following are some recommendations targeted towards address the resistance toward cloud computing:

1- Since security and privacy are a top priority for individuals and organizations, the issue of trust exists. This can be mitigated by the anointed government body actively seeking cloud computing companies in KSA and acquiring them in partnership with trusted and reputable IT firms, hence transferring the trust from a commercial brand to a government entity. Furthermore, the government can actively with the major cloud computing companies strike white-labeling deals with IT firms globally, so they can offer their solutions both in Arabic and under the governments supervision and trust.
2- Create targeted programs to recruit, train and certify technicians/engineers to serve as pool of talents to serve the ecosystem.
3- Subsidies for the costs of adopting cloud computing, preferably after white-labeling and acquiring major local cloud computing companies.

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