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Contents

Cyber Security and Digital Forensics

Data Mining

Geographic Information Systems and Image Processing

Natural Language and Speech Processing

Security and Safety Management

Software Define Network

Software Engineering and Modeling

Big Data and Cloud Computing

An Improved Differential Evolution Algorithm with Opposition-Based Learning for Clustering Problems

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Abstract

Differential Evolution (DE) is a popular efficient population-based stochastic optimization technique for solving real-world optimization problems in various domains. In knowledge discovery and data mining, optimization-based pattern recognition has become an important field, and optimization approaches have been exploited to enhance the efficiency and accuracy of classification, clustering and association rule mining. Like other population-based approaches, the performance of DE relies on the positions of initial population which may lead to the situation of stagnation and premature convergence. This paper describes a differential evolution algorithm for solving clustering problems, in which opposition-based learning (OBL) is utilized to create high-quality solutions for initial population, and enhance the performance of clustering. The experimental test has been carried out on some UCI standard datasets that are mostly used for optimization-based clustering. According to the results, the proposed algorithm is more efficient and robust than classical DE based clustering.

Keywords: *differential evolution algorithm, clustering, opposition-based learning*

I. INTRODUCTION

The rapid progress in technologies for data storage and the remarkable increase in internet applications have made a huge amount of different types of data. Undoubtedly, these data involve a lot of useful information and knowledge. Data mining is an efficient way to extract valuable hidden patterns from these large data sets. Clustering, often called unsupervised learning, is one technique for finding intrinsic structures from data with no class labels. It partitions a given dataset into different sets called clusters such that members of a cluster are more similar to each other and these are dissimilar from members in other clusters. Cluster analysis has been successfully used in various domains such as image processing, web mining, market segmentation, medical diagnosis, etc. The various kinds of clustering approaches have been proposed and used in different research communities [1]. Partition-based and hierarchy-based clustering are the two most important approaches [2]. In partition-based clustering, the data instances are divided into a given number of different partitions based on their similarities. In hierarchy-based clustering, a nested sequence of partitions is produced by representing as a dendrogram. This paper emphasizes on the partition-based clustering. The most well-known and widely used partition-based clustering approach is K-means [3]. Nevertheless, it is sensitive to initial settings and may trap to local optima. In recent times, optimization-based clustering approaches have become an attractive way in solving cluster analysis problems [4] [5] due to their population-based, self-organized and decentralized search behavior and their ability of discovering superior results.

DE is a simple, efficient and robust optimizer for many real-world global optimization problems in various domains. It has been successfully utilized as an effective alternative way to solve clustering problems [6] [7]. DE, however, sometimes fails to meet the global optimum. It may suffer from the situation of stagnation in which DE may stop searching the global optimal solution even though it has not caught the local optima. DE is vulnerable to premature convergence which may take place due to the loss of diversity in population. Besides, DE's performance relies on control parameter settings and the positions of initial population. If the initial population is composed of high-quality individuals, it is more likely to give rise to higher quality or acceptable solution. Many research works have been recently introduced to boost the efficiency of standard differential evolution algorithm [8] [9]. This paper aims to employ the DE algorithm for clustering problems. In order to boost the clustering effectiveness of DE based algorithm, a two-step population

initialization method (OBL) is utilized to create the high-quality initial population.

The rest sections are structured as follows: In section 2, a background about the basic concepts related to the canonical DE algorithm and OBL approach, and existing works related to cluster analysis utilizing DE are presented. The proposed approach is introduced in Section 3 and the conducted experimentations for evaluating the clustering performance of the DE variants are presented in Section 4. Lastly, Section 5 describes the conclusion of this paper.

II. BACKGROUND

In this section, the brief description of DE algorithm and OBL scheme, and then the related works to cluster analysis based on DE are described.

A. Differential Evolution Algorithm

Differential evolution (DE) algorithm, proposed by R. Storn and K. Price in 1995, is a simple and dominant population-based nature-inspired approach to solve global optimization problems [10]. Several variations of the DE algorithm have been recently introduced and employed to resolve optimization problems in several domains. DE, like other population-based algorithms, evolves the population of solutions at each generation by reproduction processes to deliver a better solution. The procedure of standard DE algorithm involves four successive steps such as initialization of a population, mutation, crossover, and selection. As soon as the first step is performed by generating an initial population, DE performs three remaining steps iteratively until a stopping situation. A brief description of all these steps based on the traditional DE [11] is presented in the followings.

1) Population Initialization

DE generally constructs the initial population with a set of candidate solution vectors (also called as chromosomes or individuals) that are randomly selected from the search space. Each solution vector X_i in the population, $P=\{X_1, X_2, .., X_{NP}\}$ at the t^{th} iteration is denoted as $X_{i,g}=\{x_{i,t}^1, x_{i,t}^2, \ldots, x_{i,t}^d\}$ where NP represents the number of population and d refers to the number of solution dimensions.

2) Mutation

Once the population is initialized, a mutant vector V_i is created for each parent vector X_i by perturbing a target vector with a weighted difference of two random solution vectors from the current population as stated by the following equation:

$$V_i=X_a+f(X_b-X_c) \quad (1)$$

where X_a, X_b and X_c are three randomly selected vectors such that a, b, c ∈ [1, NP] and i ≠ a ≠ b ≠ c, and then f is the scaled factor within (0, ∞).

3) Crossover

In crossover step, an offspring vector U_i is created by recombining the parent X_i and the mutant V_i. Crossover is implemented as follows:

$$u_{i,t}^j=\begin{cases} v_{i,t}^j & \text{if rand(j)}\le CR \\ x_{i,t}^j & \text{otherwise} \end{cases} \quad (2)$$

where rand(j) ∈ U(0,1), i∈[1,NP], j∈[1,d], and CR is the crossover rate within (0,1).

4) Selection

The selection phase determines the survival solution among the parent and offspring vectors according to the value of fitness function. For a maximization problem, the solution with the larger value of fitness will survive in the iteration as follows:

$$X_{i,t+1}=\begin{cases} U_{i,t} & \text{if } f(U_{i,t})>f(X_{i,t}) \\ X_{i,t} & \text{otherwise} \end{cases} \quad (3)$$

where f(.) indicates the value of fitness function.

There are numerous variants that were extended from basic DE. These variants are denoted by DE/x/y/z notation where x indicates the way of choosing a target vector; y specifies the number of pairs of vectors to compute difference vectors, and the last symbol, z indicates the recombination scheme for the crossover operator [11]. Normally, the DE algorithm's performance highly relies on control parameters, adopted mutation strategy, and population size and positions.

B. Opposition-Based Learning

An innovative scheme for machine intelligence algorithms, Opposition-based learning (OBL) has been proposed by Tizhoosh and utilized to accelerate genetic algorithm (GA), artificial neural networks (ANN), reinforcement learning [12] [13], and differential evolution algorithm [14] [15]. Numerous researches have carried out the integration of population-based optimization approaches with the OBL scheme to enhance their search behaviors. The key idea of OBL is searching for a superior estimation of the current candidate solution by considering estimates and their

respective opposite estimates together. Definition 1 and 2 are the concept of OBL described in [12].

Definition 1- If x be a real number in a range of $[x_{min}, x_{max}]$, then the opposite number of x, $\breve{x}$ can be defined as follows:

$$\breve{x}=x_{min}+x_{max}-x \quad (4)$$

Definition 2- If $P(x_1, x_2,\ldots, x_n)$ is a point in n-dimensional space such that $(x_1, x_2,\ldots, x_n)\in R$ and $x_i\in[x_{i,min}, x_{i,max}]$, then the opposite point of P, $\breve{P}(\breve{x_1}, \breve{x_2},\ldots,\breve{x_n})$ can be defined by its elements as follows:

$$\breve{x_i}=x_{i,min}+x_{i,max}-x_i \quad (5)$$

Opposition-based optimization generates an opposite solution for each candidate solution. After calculating their fitness values, the fitter one among the candidate and its opposite will survive in the evolution process. This paper uses this concept to find initial cluster solution for DE.

C. Related Works

The application of differential evolution to the cluster analysis has been an interesting research topic for a long time. The work proposed by S. Paterlini and T. Krimk [6] can be regarded as innovative effort in this field. The authors investigated the clustering performance of DE, GA, and particle swarm optimization (PSO) by employing medoid-based representation. Finally, they concluded that DE is superior compared to the other approaches and more suitable rather than GA for clustering analysis. In [16], the authors proposed a dynamic shuffled differential evolution algorithm for data clustering (DSDE) to enhance the speed of clustering convergence. This work proposed a random multi-step sampling as an initialization method due to most of clustering algorithms are sensitive to the chosen initial centroids, which can lead to premature convergence. They also applied a sorting and shuffled scheme to divide the whole population into two subpopulations in order to enhance the diversity of population. In this proposed approach, DE/best/1 mutation strategy is employed for both subpopulations during the process of evolution to exchange the direction information among the populations effectively and to balance the exploitation ability of this mutation strategy. According to the results, DSDE is superior to the classical DE and other well-known evolutionary algorithms in term of total intra-cluster distances.

The differential evolution based clustering approach with K-mean algorithm (DE-KM) [17] was proposed to catch high quality clustering solutions in term of sum of squared errors (SSE). In this work, the K-mean algorithm was incorporated into the process of DE to create the initial population and optimize the offspring solution. Their reported experimental results described that the incorporation of DE with a local search algorithm is superior to the DE only. In [18], an efficient data clustering approach based on DE was presented to manage the weaknesses of k-means algorithm. The proposed work utilized the classical DE with the within-cluster and between-cluster distances as objective functions. They concluded that their presented approach was comparable to the K-means and achieved better solutions. In [19], the author proposed a differential evolution algorithm with macromutations (DEMM) to enhance the exploration ability of the classical DE for clustering. In DEMM, the macromuation scheme was applied with the application probability and macromutation intensity that are dynamically changed during the evolution process. The application probability was used to shift between the common mutation and crossover and macromutaions. The intensity (crossover rate) of the macromutations was exponentially decreased in order to get wider exploration at the initial stage and then gradually turn into better exploitation at the later stage. As the performance of the DE algorithm depends on the adopted mutation strategies, a new DE variant, Forced Strategy Differential Evolution (FSDE) was proposed and applied it for data clustering in [20]. In FSDE, a new mutation strategy was presented, which applied two difference vectors based on the best solution vector. Besides a constant traditional scaling factor, this strategy used an additional variable control parameter. FSDE applied the result from K-means as one member of the initial population and then chose the rest of the population randomly. According to the stated results, the FSED delivered fine cluster solutions based on different cluster validity measures.

The main intention of this paper is to boost the clustering performance of the DE algorithm by adapting a two-steps initialization method. Opposition-based learning proposed to accelerate the machine intelligence algorithms will be used as an initialization method.

III. A Differential Evolution based Clustering Algorithm with Opposition-Based Learning

For applying the DE algorithm to the clustering problem, a chromosome encodes a cluster solution that is represented by vectors of real numbers. Hence, the length of each chromosome depends on the dimension of the given dataset and the number of clusters in the dataset. If K is the number of clusters and d is the number of dimensions of the dataset, K*d will be the length of the chromosome , where the first d-genes of the chromosome represents the first cluster solution, the next d- genes encodes the second solution, and the last d-genes is the K^{th} cluster solution as shown in Fig.1.

In the optimization based clustering problem, cluster validity measures are used as objective functions. In this paper the fitness of every chromosome is calculated by using the total intra-cluster distance (IntraD) which is formulated as follows:

$$\text{IntraD} = \sum_{j=1}^{k} \sum_{p \in C_j} d(p, c_j) \quad (6)$$

where k is the number of clusters, p is a data point in the j^{th} cluster C_j, c_j is the cluster center of C_j and then $d(p, c_j)$ denotes the Euclidean distance between data point and cluster center of C_j. In this work, a two-step initialization technique is adopted in order to get better positions of the initial population. The opposite-based learning is exploited to enhance the quality and diversity of initial population. To generate the initial population, each solution vector $X_i=\{x_i^1,\ldots, x_i^d, \ldots, x_i^{(k-1)d+1}, \ldots, x_i^{kd}\}$ is firstly initialized with randomly selected k data point from the given dataset. And then its opposite vector $\breve{X}_i=\{\breve{x}_i^1, \ldots, \breve{x}_i^d, \ldots, \breve{x}_i^{(k-1)d+1}, \ldots, \breve{x}_i^{kd}\}$

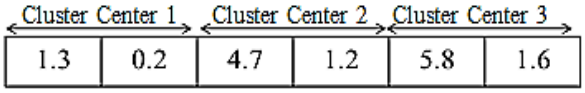


Figure 1. Chromosome Encoding for a Cluster Solution

is calculated according to the equation (5). The fitness of each solution vector and its opposite vector are evaluated and then, the fitter vector is selected as an initial solution. Once the initial population is constructed, the evolution processes are accomplished until a stopping condition is met. In Fig. 2, the proposed DE based clustering algorithm, DEC-OBL is given.

Algorithm: DEC-OBL
Input: Dataset (D), Number of clusters (k), Maximum iteration (maxIt), Number of population (NP), Scaling factor (F), Crossover rate (Cr)
Output: Cluster centers
//Population Initialization with OBL scheme
For i=1 to NP
1. Initialize a chromosome with k randomly selected data points
2. Calculate the opposite chromosome according to eq. (5)
3. Evaluate the fitness of two chromosomes according to eq. (6)
4. Select the fitter one from the pair of chromosomes

End
// Evolution Process
For i=1 to maxIt
1. Generate the mutant vector by applying the mutation operation
2. Generate the offspring vector by applying the binomial crossover operator
3. Evaluate the fitness offspring vector
4. Update the population by selection operation

End

Figure 2. Differential Evolution based Clustering Algorithm with Opposition-Based Learning

IV. Experimentation

This section provides the computational results of the proposed approach. The aim is to study the impact of population initialization technique on the clustering performance of the DE based approaches. The proposed algorithm and the traditional DE based clustering algorithm (with random initialization method) were evaluated on two mutation strategies, DE/best/1and DE/rand/1. These algorithms were implemented on Core i7 processor, 8GB RAM, and 64-bit operating system using the java programming language (NetBean IDE 8.2).

The experimental test was conducted on a number of the mostly used UCI benchmark data sets for optimization-based clustering [5]. The summary of the used datasets is presented in Table I. The control parameters were set as follows: the size of the population = 100, the number of maximum iterations = 100, the scaling factor, F = 0.9, and the crossover rate, Cr = 0.5. Each approach was independently run 30 times on each dataset.

In Table II, the obtained results are given in terms of maximum, minimum, mean and standard deviation. As reported in Table II, DEC-OBL is superior to DE on both mutation strategies. The

adaptation of the OBL based initialization method can enhance the quality of cluster solutions. Moreover, DEC-OBL achieved the smaller values of the standard deviation on all data sets. Thus, the proposed approach is more effective and robust than DE.

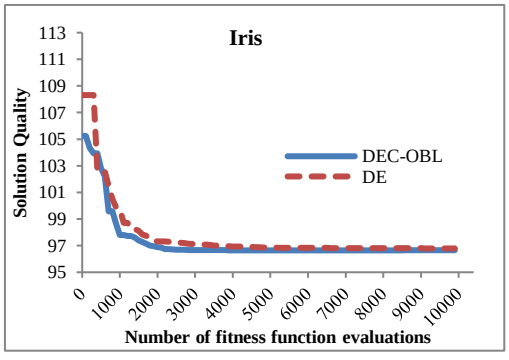
The convergence rate of DE and DEC-OBL with the DE/best/1 mutation strategy is shown in Fig. 3. As may be noticed from figure 3, the convergence rate of DEC-OBL is slightly faster than DE, and DEC-OBL achieves a better exploration of the search space at the early stage of the searching.

TABLE I. THE DESCRIPTION OF DATASETS

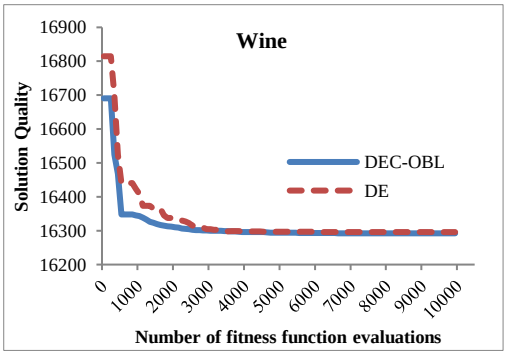
Datasets	No. of features	No. of data instances	No. of Clusters
Iris	4	150	3
Wine	13	178	3
Glass	9	214	6
Cancer	9	683	2
Thyroid	5	215	3

TABLE II. COMPARISON FOR SUM OF INTRA-CLUSTER DISTANCE OF DE AND DEC-OBL WITH TWO MUTATION STRATEGIES

Datasets	Mutation Strategies	Algorithms	Maximum	Minimum	Mean	Std.
Iris	DE/rand/1	DE	102.1521	101.475162	99.4258	0.794783153
		DEC-OBL	100.2946	97.875	99.70693	0.018106804
	DE/best/1	DE	97.4226	96.84346	96.6578	0.273315338
		DEC-OBL	96.6578	96.65593	96.6554	0.00071655
Wine	DE/rand/1	DE	16342.188	16311.866	16319.52917	9.278565097
		DEC-OBL	16312.8125	16300.004	16306.56595	4.709555761
	DE/best/1	DE	16313.771	16295.3475	16298.85414	5.527263155
		DEC-OBL	16295.817	16292.639	16294.61315	1.221286072
Glass	DE/rand/1	DE	248.6747	239.069	243.760621	3.651353923
		DEC-OBL	242.1051	235.286	239.78097	2.258780807
	DE/best/1	DE	223.29367	218.3881	220.372695	1.823091664
		DEC-OBL	217.4788	213.3792	215.410262	1.196080611
Cancer	DE/rand/1	DE	3008.2185	2989.3984	2998.99096	8.07684039
		DEC-OBL	2988.9023	2977.8376	2983.9711	3.59454961
	DE/best/1	DE	2984.6045	2965.6616	2968.92243	5.64962512
		DEC-OBL	2965.309	2964.4873	2964.72553	0.30670541
Thyroid	DE/rand/1	DE	1898.324	1892.0066	1895.27901	2.39391237
		DEC-OBL	1882.9974	1878.0023	1881.18675	1.54906751
	DE/best/1	DE	1897.0674	1869.5518	1887.54791	9.17279202
		DEC-OBL	1869.4032	1866.5364	1867.16008	1.10661843



(a)



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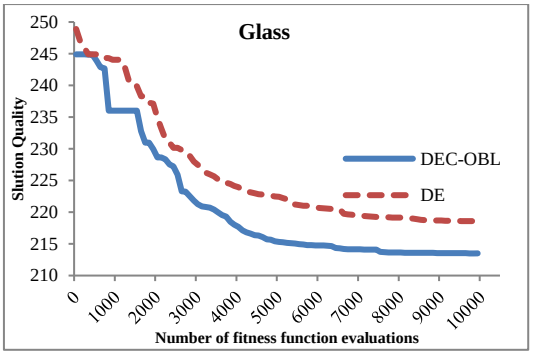

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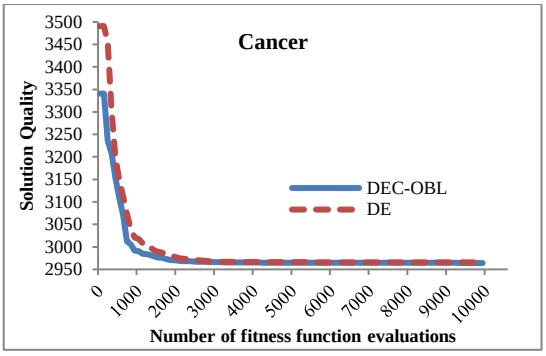

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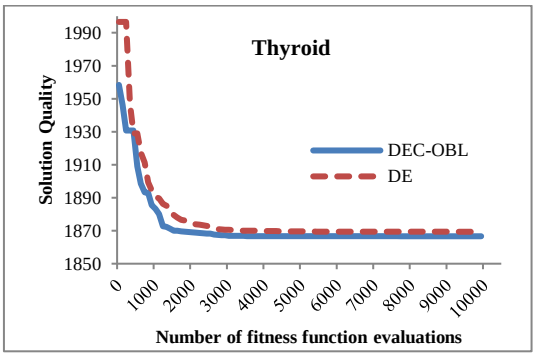

(e)

Figure 3. Convergence Performance of DE and DEC-OBL with DE/best/1 Mutation Strategy on Datasets: (a) Iris, (b) Wine, (c) Glass, (d) Cancer, (e) Thyroid

V. CONCLUSION

In this paper, a DE based clustering algorithm is presented. The idea of OBL has been exploited to enhance the clustering performance of DE. OBL is used for generating the initial solutions instead of selecting randomly. According to the obtained results, the proposed algorithm achieves the better cluster solutions and it is more robust than classical DE based clustering. As future works, the convergence speed of the proposed algorithm will be enhance by dynamically adjusting the scaling factor and crossover rate, and different cluster validity measures will be considered as fitness function. Moreover, the effect of population size on the cluster results will be investigated.

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An Improvement of FP-Growth Mining Algorithm Using Linked List

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Abstract

Frequent pattern mining such as association rules, clustering, and classification is one of the most central areas in the data mining research. One of the foremost processes in association rule mining is the discovering of the frequent pattern. To draw on all substantial frequent patterns from the sizable amount of transaction data, various algorithms have been proposed. The proposed research aims to mine frequent patterns from the sizable amount of transaction database by using linked list. In this method, first scanning the database, the count of frequent 1-itemsets is searched using the hash map and for next itemsets, it is stored in the linked list, second scanning the database. The frequent 2-itemsets is generated using hash table and so on. So, the proposed research needs only two scans and this proposed method requires shorter processing time and smaller memory space.

Keywords: *frequent pattern mining, data mining, linked list, hash table*

I. INTRODUCTION

Data mining is to draw forth the applicable information from the sizable database. The association rule mining is one of the substantial matters in the field of data mining. The frequent pattern mining is the core process of association rule mining. The frequent pattern mining, which searches the relationship in a given data set, has been widely employed in various data mining techniques. The mined information should be wide-ranging that is hidden in the data and provides some facts and information that can further be used for management decision making and process control. Several algorithms have been developed for mining frequent patterns that are significant and can provide important information of planning and control.

In frequent pattern mining, it is necessary to consider a dataset, D $\{T_1, T_2, T_3, \ldots, T_n\}$ and so, it consists of "n" transactions. Each transaction T encloses a number of items of the itemsets $I = \{i_1, i_2, i_3, \ldots, i_m\}$. Each transaction ($TID$, I) is combined together with an identifier, called TID. The minimum support count, "*min-sup"*, the percentage of transactions in D. Assume A be a set of items. If $A \subseteq T$, a transaction T is said to contain A. In the transaction set D with support s, the rule

$A \Longrightarrow B$ holds it contains $A \cup B$.

Support $(A \Longrightarrow B) = P(A \cup B)$

The rule $A \Longrightarrow B$ has confidence "c" in the transaction set D, where "c" is the percentage of transaction in D containing A that also contain B.

Confidence $(A \Longrightarrow B) = P(B \mid A)$

Association rule mining is essential in data analysis method and data mining technology. R. Srikant proposed the Apriori algorithm, which employs iterative approach and the candidate itemsets generation. If the frequent k-itemsets exist, then it scans the k times fully database. This result is more time consuming and takes more memory space.

FP-Growth algorithm was proposed by Han et al. The set of frequent 1-itemset are collected by scanning the database. And then, FP-Tree, whose structure has only frequent 1-items as nodes, is constructed. And then it stored information about the frequent patterns. This method mines frequent patterns without using the generation of candidate itemsets and the database scans twice. A set of item-prefix subtrees has in the FP-Tree. Each node in the item-prefix subtree contains three fields - item node, count, node-link.

II. RELATED WORK

The various improvements in FP-Growth algorithm have been made by the researchers. Some algorithms are discussed.

In [2], frequent pattern mining using linked list(PML) was presented. Horizontal and vertical data layout are used. For frequent 1-itemset, horizontal data layout is employed. For frequent 2-itemsets and more, vertical data layout is used. Using intersection

operations, transaction ids are speedy counted. This is the significant highlight of vertical data layout. When the frequent itemsets are large, the PML method runs faster than other methods (Apriori, FP-Growth and Eclat algorithms).

In [3], the frequent itemset mining using the N-list and subsume concepts (NSFI) was introduced by Vo, B., Le, T., Coenen, F. and Hong, T.P. The procedure of creating the N-list associated with 1-itemsets is modified and N-list intersection algorithm is improved by using hash table. Moreover, the subsume index of frequent 1-itemset based on N-list concept is determined. This method suggested two theorems. In the light dataset, NSFI did not improve over PrePost method. In the compact dataset, NSFI method is speedier than Prepost method and dEclat method.

In [4], "an improvement of FP-Growth association rule mining algorithm based on adjacency table" was proposed by Yin, M., Wang, W., Liu, Y. and Jiang, D. The items in the adjacency table were stored using hash table. In this algorithm, only one scan is needed for the transaction database, to make the input/output jobs smaller to a certain degree. Particularly, this method proves to have the high performance in the dense transaction.

In the finding of Nadi, F., Hormozi, S.G., Foroozandeh, A. and Shahraki, M.H.N., the transactions elements of database are translated into a square matrix [1]. Then, this matrix is considered as the complete graph: one-to-one correspondence, and maximum complete subgraphs are pulled out as maximal frequent itemsets. This method has fit performance in the sizable database which has particularly small number of unique items compared to the complete number of transactions.

In [5], Zhang, R., Chen, W., Hsu, T.C., Yang, H. and Chung, Y.C presented "A combination of Apriori and graph computing techniques for frequent itemsets mining: ANG". In this method, Apriori method is very efficient when frequent small-itemsets are searched. When frequent large-itemsets are searched, the graph computing method is used. So, using the advantages of two methods, hybrid method was proposed. But, in this method, the accurate switching point is essential.

III. METHODOLOGY

A. The steps of frequent pattern mining using linked list

The proposed method only needs twice to scan the database.

Firstly, the frequency of 1-itemsets is counted scanning the database and removed this 1-itemsets that the support count(Sup-count) is less than minimum support count(min-sup).

Secondly, each of frequent 1-itemsets is stored in hash table as the key.

Thirdly, each of transaction is sorted decreasing order over frequent 1-itemsets and for the next itemsets, it is stored in the linked list.

Finally, the 1-itemsets related the key are counted with sup-count and removed this 1-itemsets whose frequency is less than min-sup and so on.

B. The algorithm for frequent itemsets generation

Algorithm: Find frequent itemsets using linked list
Input:

- *D*, a database of transactions;
- *min-sup*, the minimum support count

threshold.
Output: frequent itemsets in *D*.
Method:
(1) Count the number of 1-itemsets from the transaction by scanning the database.
(2) Find the frequent 1-itemsets that satisfy min-sup.
(3) if (frequent 1-itemsets exists),
{ Set frequency = true.
Output frequent 1-itemsets.
Go to step 4.}
else
{ Set frequency = false.
Output "No frequent 1-itemsets".
Break.}
(4) Create Transaction Linked List hash table with items found in step-2 as hash table key.
(5) For each transaction of the database (//Scan D)
{-Sort the decreasing order with the items found in step-2.
Search the node related Transaction Linked List hash table key for the following itemsets in the database.
If exists, the frequency of this itemsets increases by one.
Else, create the node of this itemsets and the frequency sets one.
}
(6) Initialize k = 2.
(7) While (frequency)
{
If (k = = 2), then

{
Create 2-Itemsets Linked List hash table using the key in the Transaction Linked List hash table.
Count the frequency of 1-itemsets related the key using Transaction Linked List hash table.
Prune the 1-itemsets whose frequency is less than min-sup and update 2-Itemsets Linked List hash table
}
Else
{
Create k-Itemsets Linked List hash table using the key of k-1 Itemsets Linked List hash table.
Create the node of k-1 itemsets related the key using k-1 Itemsets Linked List hash table and then count the frequency of this node using Transaction Linked List hash table.
Prune the node of k-1 itemsets that the frequency is less than min-sup and update k-Itemsets Linked List hash table.
}
-if (frequent k-itemsets exists), =
{
Set frequency = true.
Output frequent k-itemsets.
}
else
{
Set frequency = false.
Output "No frequent k-itemsets."
}
Increases k;
}

As an example, the database shown in table 1 is explained and predefined minimum support count(min-sup) is 3.

Table 1: Transaction Database

Transaction	Items
T1	a, d. f
T2	a, c, d, e
T3	b, d
T4	b, c, d
T5	b, c
T6	a, b, d
T7	b, d, e
T8	b, c, e, g
T9	c, d, f
T10	a, b, d

Firstly, the frequency of each item is counted with the support count(Sup-count) by scanning the database shown in table 2.

Table 2: frequency of each item

1-itemset	Sup-count
a	4
b	7
c	5
d	8
e	3
f	2
g	1

And then, the frequent 1-itemset are collected by pruning the 1-itemset whose frequency is less than minimum support count that are shown in table 3.

Table 3: frequent 1-itemset

Frequent 1-itemset	Sup-count
a	4
b	7
c	5
d	8
e	3

Each of frequent 1-itemset is stored as the key in the Transaction Linked List hash table shown in figure:1 and by scanning the database, each transaction of the database is sorted in decreasing order based on frequent 1-itemset shown in table 4 and the count of the next itemsets that are related with the hash table key are stored in the Transaction Linked List hash table.

Table 4: sorted transaction

Transaction	Items
T1	d, a
T2	e, d, c, a
T3	d, b
T4	d, c, b
T5	c, b
T6	d, b, a
T7	e, d, b
T8	e, c, b
T9	d, c
T10	d, b, a

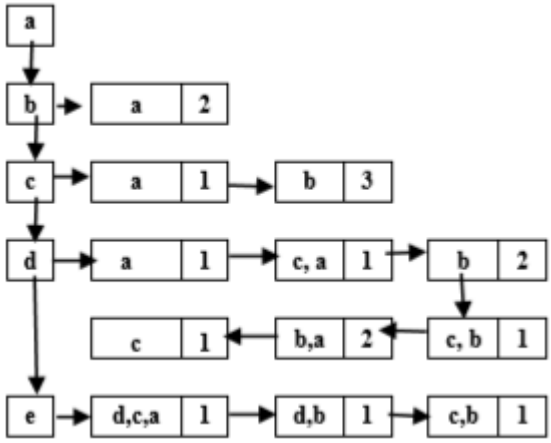

Figure:1 Transaction Linked List hash table

In addition, the 2-Itemsets Linked List is created using the key of the Transaction Linked List and then one itemsets that are related the key are counted shown in figure:2.

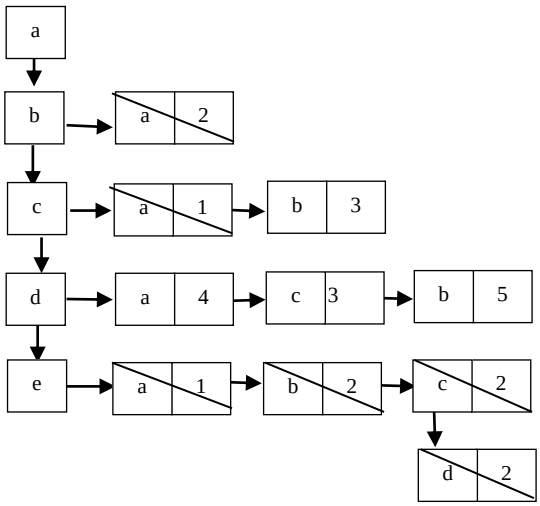

Figure 2: 2-Itemsets Linked List hash table

In accord with this min-sup count, it is necessary to get rid of infrequent items and update 2-Itemsets Linked List shown in figure:3.

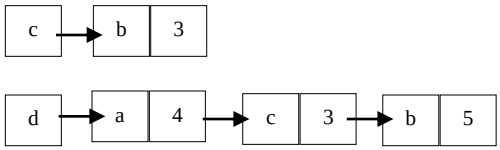

Figure 3: Updated 2-Itemsets Linked List hash table

Table 5: frequent 2-itemsets

Frequent 2-itemset	Sup-count
b, c	3
a, d	4
b, d	5
c, d	3

The frequent 2-itemsets are {b, c: 3, a, d: 4, b, d: 5, c, d: 3}.

And then, it is to create 3-Itemsets Linked List hash table shown in figure:4 using the key of 2-Itemsets Linked List and the two itemsets related the key are counted in the Transaction Linked List and pruned the itemsets whose frequency is less than min-sup count. Similarly, three itemsets are counted in the 4-Itemsets Linked List and pruned and so on.

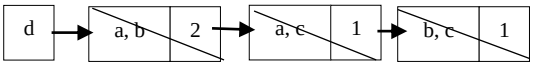

Figure 4: 3-Itemsets Linked List hash table

If there is no frequent itemsets, the process has stopped. In this example, frequent 3-itemsets does not occur because the frequency of next two itemsets is less than min-sup. So, the processing has stopped and frequent 1-itemsets and 2-iemsets are generated as the output.

IV. CONCLUSION

After having studied the mining process of frequent patterns algorithm, this paper proposes an improved FP-Growth method based on linked list. In the proposed algorithm, the database is scanned only twice and this tremendously reduces the input/output operations. The hash table is adopted in this algorithm for the speedy lookup. The proposed method has considerably reduced the running time and used up less memory space. The future work will be able to perform the comparison of the improved FP-Growth method employing linked list and FP-Growth.

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Community Detection in Scientific Co-Authorship Networks using Neo4j

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Abstract

Community structure in scientific collaboration network has become an important research area. Co-author of a paper can be thought of as a collaborative document between more than one authors. Community detection in co-authorship network reveals characteristic patterns of scientific collaboration in computer science research and help to understand the identity-organization of the author community. Louvain algorithm is a simple, easy to implement and efficient to recognize community in huge networks. In this paper, it is used to examine the structure of community in Computer University's coauthor network in Myanmar. Neo4j is also used to visualize the co-authorship network analysis results. Modularity is used to measure the quality of the cluster structure found by community discovery algorithms. In experiment, Louvain algorithm gives more effective qualitative community structures than other algorithms in co-authorship network.

Keyword: *co-authorship network, community detection, modularity, Neo4j*

I. INTRODUCTION

Co-authorship relationship is well documented form and one of the most visible of scientific collaboration. Co-authorship networks analysis is used to rank most influential authors in a co-author network, or to estimate future research collaboration, to determine the most appropriate reviewers for a manuscript. [1]. Research collaboration and co-authorship is science in an interesting multi-faced phenomenon. In co-authorship network, nodes represent paper's authors, and two authors are connected by a relationship in which they published at least one research paper.

Co-authorship social network is one kind of the social relationship network. A community is a cluster of a network where internal nodes connections are closer than external nodes. Detection community in social relationship network help to understand the network structure, to identify subgroup and to visualize the result communities. Community detection in social networks are limited and no ground truth solution to compare and it may have more than one solution to the problem. Researcher have been developed many kinds of community detection algorithms to detect community in social networks.

Bibliographical record of University of Computer Studies, Mandalay are collected from UCSM research repository [2]. The UCSM research repository is an open access institutional repository that provides search access to research publication written by UCSM staff and students. Community detection in UCSM co-authorship network reveals type of scientific collaboration in the university. It also helps us to understand the own-community of UCSM community authors. Next several years, UCSM co-authorship will emerge. At this time, current time community form can be compared with overtime community form. Co-authorship community structure can get many advantages such as collaborative work among research increases research productivity both in terms of quality and quantity of publication. Sometime, two or more researchers who have different community should collaborate to emerge new trend or technology. The purpose of this paper is to provide overview of growing research on co-author network approach to research collaboration, identifying gaps for future research.

Neo4j Desktop is a convenient way for developers to work with local Neo4j database. Graph algorithm in Neo4j library is installed as plug-in and launched the Neo4j Desktop. Louvain algorithm in Neo4j's graph algorithms is used to detect graph community structure. It is one kind of community detection algorithm that relies upon a heuristic for maximizing the modularity [3]. In this paper, the communities structure of UCSM co-authorship network is detected by using Louvain community detection algorithm in Neo4j. According to the previous research findings, it has been used with success for many kind of networks and it is suitable for more than hundred million node and billions of edges. It has been the most widely used method for detecting communities in large networks [4].

The next parts of the paper are described as follow. The theory background of network

community detection algorithms is described in section II. In the next sections, section III explores the literature reviews and section IV presents the dataset collection. Section V is the experimental results and section VI gives conclusion.

II. BACKGROUND THEORY

In this section, the necessary background knowledge of network community detection, modularity structure, co-authorship network, community detection algorithm and about Neo4j are presented.

A. Community Detection

Formation of communities is common in all types of networks and identifying them is essential for evaluating group behaviour and emergent phenomena. This information helps to infer the behaviour or similar preferences of peer groups, assess flexibility, find nested relationships, and provide information for other analysis. Community detection algorithms are also commonly used to produce network visualization for general exploration [5]. A community is a subgroup of a network where internal connections are part of a denser network than external connections. Detecting communities helps to understand network structure, to identify cohesive sub-cluster, and to draw a readable network's map.

The method of detecting by the community is to partitions which maximize connection density within a group by taking into account the connection density between groups and finding dense optimal sub-graphs in large graphs. Many community detection algorithms have been developed to find optimal communities in reasonable fast time.

B. Modularity

There are many kind of evaluation metrics to measure the quality of result community. Evaluation metrics are based on internal connectivity, external connectivity, consider on both internal and external connections and model of network.

Modularity Q is a model of network based quality metric. It is used to measure the quality of partitioning the network into cluster detected by the proposed algorithm. It is the measure of the density of intra-community relationship as compared to inter-community relationship [6]. Modularity Q can be calculated by using equation 1.

$$Q = \frac{1}{M} \sum_{ij} [A_{ij} - \frac{d_i d_j}{M}] \delta(c_i, c_j) \qquad (1)$$

where M=2e, e is the number of edges, A_{ij} is 1 if there exists the actual number of edge between i and j else it is 0, d_i is the degree of node i, d_j is the degree of node j, c_i is the cluster of i, c_j that of j and δ function is 1 if i and j reach in same cluster and 0 otherwise.

Step by step calculation of modularity are described as followed. Input of given network is edges list text file. Example network structure's edge lists are (1—2, 1—3, 2—3, 3—4, 4—5, 4—6, 5—6). Louvain method detects two community structure for the example network. One community contains node 1, node 2 and node 3 (C1= {1,2,3}) and next community contains nodes 4,5 and 6 (C2={4,5,6}). To calculate modularity Q, M=2*7, A_{ij} is the adjacency matrix of each node (nodes defined by the row and the column have different element values) and $\delta(c_i, c_j)$ is 1 if node i and node j are in the same community. Delta function use the result community structures C1={1,2,3} and C2={4,5,6}. After replacing the corresponding values in equation 1, modularity Q will be produced. In this example, Q is 0.3571429.

Q's value lies in the range [−1, 1]. For most real-networks' result community structures, the value of Q is above 0.3. Higher modularity values imply strong community structure. Most of the traditional and proposed community detection algorithms aim to optimize modularity value.

C. Co-Authorship Network

Co-authorship network is two authors have a connection if they write together at least one paper. Node in a co-authorship network represents author in which author published at least one paper. The network of co-author is one of the most tangible and well documented forms of scientific collaboration. Co-authorship network analysis is useful in understanding the structure of scientific collaborations and individual author's status.

The co-authorship network is represented as a graph G= (V, E). In which, the V is the set of researcher and E is the set of relationships if two researchers have co-authored a paper together. The primary application of co-authorship networks is to study the structure and evolution of scientific collaboration. Figure 1 shows the sample visualization of co-authorship network using Neo4j. In this network, author Aye Aye and author Mya Mya wrote a paper together and Soe Soe and Aye Aye also wrote a paper together. Analysis of the co-author network reveals characteristics of the academic community

that can help us to understand the collaboration research works and to identify prominent researchers [7].

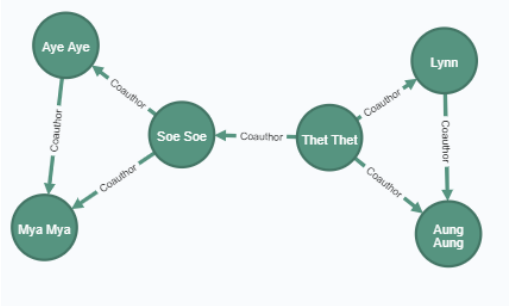

Figure 1. The sample co-authorship network(S)

D. Neo4j

Neo4j is world's learning graph database management system developed by Neo technology, Inc. It is designed to optimize fast management, storage and traversal of nodes and relationships. Graph visualization takes the understandable features one step further by drawing the graph in variety of formats, making it easier for users to work with the data. Neo4j has two visualization tools called Neo4j Browser and Neo4j Bloom that are built and designed to work with data in Neo4j's graph database. In this paper, Neo4j Browser is used for graph visualization [8].

Cypher is Neo4j's graph language and it can easily express graph structure. Graph algorithm library is installed in Neo4j Desktop. Neo4j graph algorithms help in several areas such as a route search to find the shortest route or evaluate the availability and quality of routes; determine the centrality about the importance of separate nodes and uncovering community that evaluates how groups are clustered or determines the importance of separate nodes in the network. In this paper, Louvain algorithm is used to detect community structures for co-author networks.

E. Community Detection Algorithms

There are many conventional community detection algorithms. Most of the algorithms get effective results in small and middle scale networks. For large scale networks, researcher have proposed various kind of effective community detection algorithm. They purposed to solve the scalable and time complexity challenges.Among them, some community detection algorithms are discussed in this section.

Edge-betweenness algorithm [9] finds edges connection separate modules have high edge betweenness as all the shortest paths from one module to another must pass through them. It performs by calculating the edge betweenness of the graph, removing the edge with the highest edge betweenness score, then recalculating edge betweenness of the edges and again removing the highest score edge. It returns a community object.

Infomap algorithm [10] uses random walks to analyses the information flow through a network. It is assumed that a random walker will enter the community if he spends some time traveling through the nodes of the community.

Label propagation algorithm [11] uses the information of the neighborhood node to identify community structure. Every node is labeled with their own value. Then the label of each node are replaced with the most spread in its neighborhood. This process is repeated until one of several conditions is met or no label change. The last label value is the result communities or size of communities.

Leading eigenvector algorithm [12] applies the eigenvalues and eigenvector of modularity matrix. Firstly, it calculates the leading eigenvector of modularity matrix then the graph is divided into two group. Modularity improvement is maximized depend on the leading eigenvector. In each sub division of a network, modularity is calculated. This process is repeated until the satisfy modularity result.

Louvain algorithm adopts one kind of hierarchical method called agglomerative. Each node owns a unique community. Then nodes are assigned to the community which achieve the higher modularity result and merge the communities. This process terminates when there exists only node or modularity value can't improve. Louvain algorithm is one of the fastest community detection algorithm and works well with large graphs. The advantage of Louvain is to minimize the time of computation. This mathematical method has become quite popular and consists in calculating a number of each partition which quantifies the quality of the partition and then finding the maximal modularity partition. Louvain algorithm can be called in Neo4j Desktop software when installing Neo4j's graph algorithm library.

III. Literature Reviews

Communities in social graphs may indicate groups of people with common interest. Most conventional community detection techniques are based graph partitioning, hierarchical clustering and modularity optimization algorithm [13]. Graph partitioning algorithm divides the graph into

predefined size. So graph partitioning based algorithm need to know the number of community. Hierarchical clustering techniques are based on the vertex similarity measure. These techniques don't need a predefined size and number of communities.

There are two types of categories: agglomerative and divisive algorithms. Agglomerative algorithm is bottom up approach. It starts with each node as a separate cluster and iteratively merged them based on high similarity. Divisive algorithm is top down techniques. It starts with the entire network as a single cluster and iteratively splits it by eliminating links joining nodes with low similarity and ends up with unique communities. Modularity optimization techniques is based on the modularity value to get quantitative community structure. The larger the modularity value, the better the partition.

IV. Dataset Collection

Co-authorship network has been constructed from the publication list of UCSM research repository in "https://www.ucsm.edu.mm/ucsm/". These data are crawled by web crawler and then collect coauthors information. Coauthor information can be gotten from their publication lists. Figure 2 is the publication information on the UCSM webpage. This data is collected at November, 2019 from UCSM webpage. When creating UCSM co-authorship network, it contains 80 authors and 189 relationships between authors. In this network, if a paper is write only one author, this publication will be ignored. If one publication has three author a, b and c, its relationships will be a-b and a-c [14]. Co-authorship network is constructed based on UCSM research repository's publication, and then visualize and analysed with Neo4j.

V. Experimental Analysis

The experiments are implemented on a laptop with Core i7, 8GB of RAM, 64-bit Window operating system and using neo4j-desktop-offline-1.2.3-setup. Neo4j is used for the co-authorship network analysis. Sample co-authorship network (S) that are shown in figure 1, is used as the example to detect the community. It contains six nodes and seven edges. Firstly, create six nodes and edges using the following cypher language.

Create Node:

```
MERGE (ThetThet:Author{id:"Thet Thet"});
MERGE (Lynn:Author{id:"Lynn"});
```

Create Edge:

```
MATCH (a:Author), (b:Author)
WHERE a.id = "Thet Thet" AND b.id = "Lynn"
CREATE (a)-[: CoAuthor]->(b)
RETURN a,b;
```

After creating graph structure, detect the community structure of network by using Louvain algorithm in Neo4j graph library. The cypher query for detecting community is described as follow.

```
CALL algo.louvain.stream("Author", "CoAuthor", {})
YIELD nodeId, community
MATCH (user:Author) WHERE id(user) = nodeId
RETURN user.id AS user, community
ORDER BY community;
```

Louvain community detection algorithm produces two community structures for sample network S that are shown in Table I. According to result table, {Thet Thet, Aung Aung, Lynn} are in the same group and {Aye Aye, Mya Mya, Soe Soe} are in the same group. Then, the quality of result communities is measured using the modularity Q. Modularity value gets 0.3571429.

No.	Title	Subjects	Date	Type	Publication Title	Authors	Communities
1	Feature Selection to Classify Healthcare Data using Wrapper Method with PSO Search	Data Mining and Machine Learning	2019-09-08	Journal Article	International Journal of Information Technology and Computer Science(IJITCS)	Thinzar Saw [3] , Phyu Hninn Myint [6]	Data Mining and Machine Learning Lab, Faculty of Computer Science
2	Analysis on Skin Colour Model Using Adaptive Threshold Values for Hand Segmentation	Image and Signal Processing	2019-09-08	Journal Article	International Journal of Image, Graphics and Signal Processing(IJIGSP), Hong Kong, 2019, Volume 11, No-9, pp. 25-33	Phyu Myo Thwe [2] , May The` Yu [6]	Image and Signal Processing Lab, Faculty of Information Science
3	Feature Representation and Feature Matching for Heterogeneous Defect Prediction	Software Engineering	2019-08-07	Conference Paper	International Conference on Intelligence Science (pp. 1-14). Springer, Cham.	Thae Hsu Hsu Mon [1] , Hnin Min Oo [4]	Software Engineering Lab, Faculty of Information Science

Figure 2. UCSM's research repository page

From UCSM's co-authorship network, not only network's community information but also other co-author information can also be extracted. Author and his or her co-author information are extracted from graph database. Table II shows the query result about the co-authors who worked together. In this table, Author Thet Thet wrote publication papers together with Soe Soe, Lynn and Aung Aung. If the paper is written with co-author, it will provide more value, wisdom, and improve research efficiency. Figure 3 shows the graph based structure for the author Thet Thet and her coauthors using cypher query language.

TABLE I. COMMUNITY STRUCTURE OF SAMPLE CO-AUTHORSHIP NETWORK

"User"	"Community"
"Thet Thet"	0
"Lynn"	0
"Aung Aunt"	0
"Soe Soe"	1
"Aye Aye"	1
"Mya Mya"	1

TABLE II. CO-AUTHORS' INFORMATION OF SAMPLE CO-AUTHORSHIP NETWORK

"Author"	"Coauthor"
"Thet Thet"	["Soe Soe", "Lynn", "Aung Aung"]
"Aye Aye"	[Mya Mya]
"Lynn"	["Aung Aung"]
"Soe Soe"	["Mya Mya", "Aye Aye"]

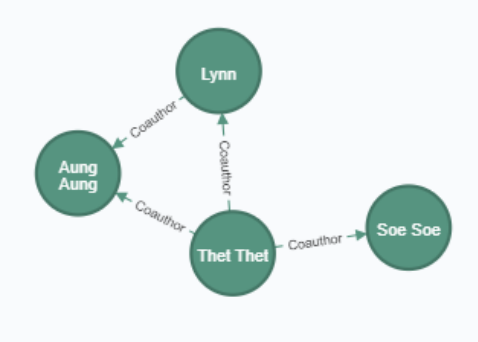

Figure 3. Specific author "Thet Thet" and her relationship

By using sample co-authorship network S, various type of co-authorship network analysis is described. In Table III, each author's publication information is extracted in Neo4j graph databased and shows with table structure. Next experiment is tested on UCSM's co-authorship network. Firstly; this graph is created and stored in Neo4j graph database. The graph structure of the UCSM co-authorship network is shown in figure 4. Current time, this network is a disconnected network between some authors and the network will emerge as large scale network at the coming years. Louvain method in neo4j graph library is used to detect the community structure in that co-authorship network.

TABLE III. EACH AUTHOR'S PUBLICATION INFORMATION

"Name"	"First Author Publication Count"	"Second Author PublicationCount"
"Thet Thet"	3	0
"Aunt Aung"	0	2
"Aye Aye"	1	1
"Mya Mya"	0	2
"Lynn"	1	1
"Soe Soe"	2	1

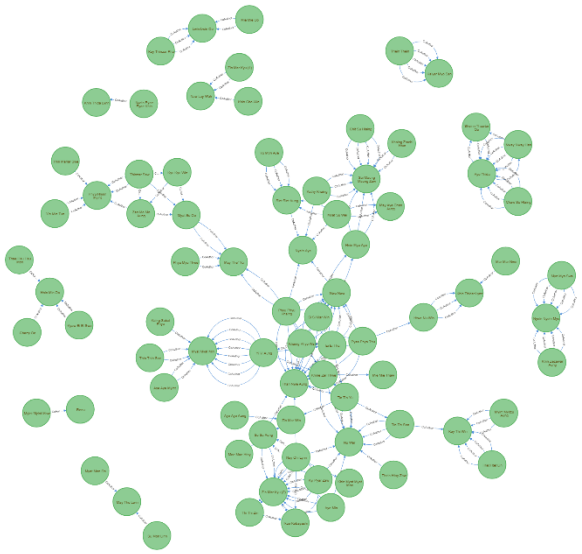

Figure 4. UCSM co-authorship network

Table IV is the result community structures of UCSM co-author network. In table IV, "Author" column is the group of member name in each community and "Community" column is only the identify group label for each group. The table describes the coauthor community structure of UCSM's co-authorship network. When learning the result communities, members in same community have similar research field and they interest in the

same area. Most influence member in each community is the supervisor or head of its research area. He or she have many relationships to other members in the same community. If authors in same community collaborate the new research, these finding can improve. When two different communities work together, new technology trend will be emerged. Carlos Medicia Morel et.al presented that co-authorship network analysis could help address, providing a substantial contribution to global health [15]. After getting community results found by the proposed community detection algorithm, the result community's quality is measured by using Modularity Q in equation 1.

TABLE IV. Result community structures of UCSM's co-authorship network

"Author"	**"Community"**
["Than Nwe Aung", "Yi Yi Aung", "Nwe Nwe", "Phyu Phyu Khaing", "Ei Ei Thu", "Khaing Phyo Wai", "Tin Tin Yu", "Pyae Phyo Thu", "Khine Zar Thwe", "Mie Mie Thaw"]	0
["Myat Myat Min", "Aye Aye Myint", "Thin Thin Soe", "Kaing Sabai Phyu"]	1
["Aye Thida", "Nway Nway Han", "Chaw Su Hlaing", "Sheinn Thawtar Oo"]	2
["Zin Mar Kyu (2)", "Nu War", "Kyi Pyar Zaw", "Nay Chi Lynn", "Su Su Aung", "Aye Min", "Khin Myat Myat Moe", "Thi Thi Zin", "Ikuo Kobayashi", "Thein Htay Zaw"]	3
["Sai Maung Maung Zaw", "Si Si Mar Win", "Sint Sint Aung", "Hnin Mya Aye", "Chit Su Hlaing", "Myat Su Wai", "Yu Mon Aye", "Kathy Khaing", "Nyein Aye", "Khaing Zarchi Htun", "May Aye Chan Aung"]	4
["Nyein Nyein Myo", "Khin Zezawar Aung", "Myo Myo Swe"]	5
["Kay Thi Win", "Tin Tin San", "Htet Htet Lin", "Myint Myitzu Aung"]	6
["Phyu Hninn Myint", "May The` Yu", "Thinzar Saw", "Myat Su Oo", "Yin Min Tun", "Zan Mo Mo Aung", "Phyu Myo Thwe", "Thiri Marlar Swe", "Kyu Kyu Win"]	7
["Lwin Lwin Oo", "Kay Thinzar Phu", "Mie Mie Oo"]	8
["Thein Thein", "Kalyar Myo San"]	9
["Hnin Min Oo", "Kyaw Ei Ei Swe", "Thae Hsu Hsu Mon", "Cherry Oo"]	10
["Htwe Nu Win", "Mar Mar Nwe"]	11
["Nyein Pyae Pyae Khin"], ["Khin Thida Lynn"]	12
["Naw Lay Wah", "Zin Mar Kyu (1)", "Khin Cho Win"]	13
["Zin Mar Win", "Aye Aye Aung", "Moe Htay"]	14
["May Thu Lwin", "Myat Mon Oo", "Su Mon Linn"]	15
["Myint Myint Maw", "Renu"]	16

Most of data mining and machine learning tasks contain training data and testing data information. Social network analysis task does not have ground truth solution to the problem. So, the quality of result is tested using quality metrics. If the given network has ground truth result, Normalize Mutual Information can be used [16]. It compares the similarity between the detected community and ground true community of the same network. In this paper, there is no ground truth community result for UCSM co-author network. So, modularity is used to test the quality of result community.

To compare the quality of Louvain algorithm in Neo4j's graph library, four kinds of community detection algorithm such as Edge-betweenness algorithm, Infomap algorithm, Label-propagation algorithm, Leading-eigen algorithm are used. Table V shows comparison of quality metric, modularity for five algorithms on UCSM's co-authorship network. In table V, modularity Q column is modularity value of the result community structures of each algorithm and Number of Community (NC) column is the number of cluster for given network that is just knowledge about the cluster information.

According to the Table V, all of the algorithm get suitable modularity result. Among them, Louvain algorithm get highest modularity value than other algorithms. This means that the quality of result community structure using Louvain algorithm is better than the other algorithms.

TABLE V. COMPARISON OF FIVE COMMUNITY DETECTION ALGORITHMS BASED ON MODULARITY VALUE

Algorithm	**Modularity (Q)**	**Number of Community(NC)**
Edge_Betweenness	0.7862182	19
Infomap	0.6067019	26
Label_prop	0.7725848	20
Leading_eigen	0.7532404	16
Louvain (Neo4j)	0.7912292	17

Researcher comprise as a community that entrance to success in this research field. Paper publication information of researchers are also visualized using Neo4j. Ten author's publication information in the co-authorship network is described in Table VI.

TABLE VI. SAMPLE TEN AUTHORS' PUBLICATION INFORMATION

"Name"	"First Author Publication Count"	"Second Author Publication Count"
"Nay Chi Lynn"	6	0
"Nway Nway Han"	4	2
"Tin Tin Yu"	4	2
"Si Si Mar Win"	6	0
"Pyae Phyo Thu"	6	0
"Khine Zar Thwe"	4	2
"May The` Yu"	0	6
"Chaw Su Hlaing"	6	0
"Khin Zezawar Aung"	4	0
"Myo Myo Swe"	5	0

Visualization is a main component for social network analysis. It gives meaning to the analysis and both complement each other. Today, Neo4j Browser is being increasingly used to visualize co-authorship network. Neo4j can be used to run complex queries. According to the above experiment, when using Neo4j user can find most influential member in this community, automatically determine the most appropriate reviewer for manuscript or to predict future research collaboration and can understand structure and evolution of the related academic society.

VI. CONCLUSION

Community detection in social relationship network has become important research area in social network analysis. Co-authorship social network is used to detect the community in co-author network. The main contribution of this paper is that the strongly connected component of co-authorship graph own clear community structure. Author belonging to the strongest connected components are grouped into no-overlapping cohesive subgroup. One of the useful tools for graph clustering and visualization, Neo4j is used for co-authorship network analysis. Neo4j is suitable if data is represented using the graph model such as network and can run complex queries. In experiment, Louvain algorithm in Neo4j's graph library can detect more effective qualitative community structures than other algorithms.

The main drawback of Louvain algorithm is resolution limit. Resolution limit is a phenomenon in which communities that are smaller than a scale, are not specified. So, future research direction aims to propose effective and efficient improve Louvain algorithm for finding community structures on large co-authorship networks. Not only modularity but also other kind of quality metrics will be used to evaluate the community structure results. It will be implemented using Spark framework and Neo4j.

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Effective Analytics on Healthcare Big Data Using Ensemble Learning

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Abstract

Healthcare big data is a collection of record of patient, hospital, doctors and medical treatment and it is so large, complex, distributed and growing so fast that this data is difficult to maintain and analyze using some traditional data analytics tools. To solve this difficulties, some machine learning tools are applied on such big amount of data using big data analytics framework. In recent years, many researchers have proposed some machine learning approaches on healthcare data to improve the accuracy of analytics. These techniques were applied individually and compared their results. To get better accuracy, this paper proposes one machine learning approach called ensemble learning, in which the results of three machine learning algorithms are combined. Soft voting method is used for combining accuracies. From these results, it is observed that ensemble learning can obtain maximum accuracy.

Keywords: *Ensemble learning, big data analytics, soft voting*

I. INTRODUCTION

Historically, healthcare industry is highly data intensive and large amount of data generated by healthcare industry is driven by record keeping and regulatory requirements for caring patient. Most of these healthcare data are in the form of hard copy and they are required to be digitized rapidly into digital format. These big amount of healthcare data can be used to improve the quality of healthcare and also to reduce the costs. These data can be used in wide area of medical fields and healthcare functions, health management and also disease surveillance.

One of the main purpose of healthcare services is to get the best cares or services to patients. Nowadays many organizations of healthcare services proposed many models of information system. Electronic health records (EHRs) and large amount of complex biomedical data are used to get personalized, predictive and preventive medicine. Genomics and post-genomics technologies can produce large size of raw data in the process of complex biochemical regulatory for the living creatures [4]. Since these EHRs data are heterogeneous, they must be stored in different data forms, different styles and different storage types. Such data can be unstructured, semi-structured or structured. They may also be discrete or continuous.

Healthcare sector needs to be modified and modernized with the big data analytics because big data techniques is new emerging technology. Careful analyzing of healthcare big data is required to be used big data techniques in healthcare inductor. Big data are difficult to analyze and manage with traditional computations. There are huge amount of data in healthcare such as list of patients, doctors and medicine, history of patient records. Big data analytics can be used for integration of heterogeneous data and data quality control. It can also be applied in analysis, modeling, integration and validation [5]. Comprehensive knowledge discovery from large amount of data can be provided in big data analytics application. Big data analytics will discover new knowledge and it can be provided for benefit to the patients, health workers and healthcare policy makers [7].

Big data analytics for healthcare and medical field can enable analysis of large amount of datasets of thousands of patients and correlation between these datasets. This analytics can also integrate the results of analysis of many scientific areas such as bioinformatics, medical imaging, health informatics and sensor informatics.

Using machine learning techniques on big data analytics tools can enhance the performance of the techniques. Many researchers have proved these techniques individually and then compared such results with other methods. This paper proposes an ensemble learning techniques for better accuracy on healthcare big data. A big data analytics tool, Spark, is used in this proposed system. With Spark MLlib, different machine learning tools such as Naïve Bayesian, Decision Tree and K-NN are used as base learners to get particular accuracy of each method and

ensemble learning method is applied on the same dataset to get more accuracy than particular methods.

This paper is organized as follows: Section 2 describes the related research works. Characteristics of big data and some of big data analysis techniques are described in Section 3. Ensemble learning proposed in this paper is also included in this section. The next section, Section 4, includes the experimental results of this research work. Section 5 is the last section and it concludes the paper with conclusion and future works.

II. RELATED WORKS

Many machine learning algorithms have been applied in big data analysis and also healthcare big data. The authors in [7] proposed a framework for analysis of stock markets with machine learning algorithms. In this paper, forecasting on decision of stock trading was proposed using ANN and decision support model. Such decision was compared with other methods such as Naïve Bayes, SVM, K-Nearest Neighbor and Decision Tree Model.

In the paper [9], the authors proposed an algorithm called Ensemble Random Forest Algorithm to be analyzed on big data. It presented the difficulties of modelling the insurance business data with classification because of imbalanced of business data that was missing by user features and many other reasons. Heuristic bootstrap sampling approach was combined with the ensemble learning algorithm for mining on insurance business data with large-scale. Ensemble random forest algorithm was also proposed and it can be applied in the parallel computing process and Spark tool was used to optimize memory-cache mechanism. The performance of the proposed algorithm was evaluated by F-Measure and G-Mean. Its experimental results of this proposed system showed that it outperformed in both performance and accuracy with imbalanced data than other classification algorithms.

In the paper [2], the authors proposed efficiency and reliability classification approach for diabetes. The real data was collected from Sawanpracharak Regional Hospital, Thailand and this data was analyzed with gain-ratio feature selection. Naïve Bayesian, K-nearest neighbors and decision tree classification were used as base learners on the selected features. To apply the ensemble learning on these three algorithms, bagging and boosting were combined. Comparison of results of base learners and ensemble learnings were presented. Then the results of each ensemble learning with respective base learner were collected and compared to find the best method for its research work.

The EC3 ensemble learning was proposed in the paper [8]. In which, step by step processing of a novel algorithm named EC3, Combining Clustering and Classification for Ensemble Learning, was presented. Classification and clustering have been successful individually but they had their own advantages and limitations. The author proposed systematic utilization of both of these types of algorithms together to get better prediction results. Its proposed algorithm can also handle imbalanced datasets. 13 datasets from UCI machine learning repository were used and 60% was for training, 20% for testing and other 20% for validation. Six algorithms were used as base classifiers namely Decision Tree, Naïve Bayes, K-nearest neighbors, Logistic Regression, SVM and Stochastic Gradient Descent Classifier. Base clustering methods were DBSCAN, Hierarchical, Affinity, K-Means and MeanShift.

Big data tool applied on healthcare analytics was presented in the paper [5]. K-means clustering techniques was applied on healthcare big data and MongoDB was used as data storage. History data of patient's medical treatment were clustered according to their attribute values. This paper proved that using machine learning tools on big amount of healthcare data is efficient for patients, doctors and medical treatment.

III. ENSEMBLE MODEL FOR BIG DATA ANALYTICS ON HEALTHCARE DATA

Data mining has many specialized forms and machine learning is one of such specialized forms. In machine learning, models are learnt by supervised or unsupervised learning. A mathematical model is generated from a set of data in supervised learning and it includes both the inputs and the respective outputs. Classification and regression algorithms are the type of supervised learning. Classification algorithms can be used for the outputs that are restricted to a limited set of values. Regression can be used for the continuous outputs such as length and temperature. In unsupervised learning, a mathematical model is generated from a set of data. This model has only inputs and no desired output labels. It can be used to search structure in data, such as data points clustering. This type of learning can find patterns in data, and the input can be grouped into clusters or categories, as in feature learning. The process of reducing the number

of features and dimensionality reduction can be applied on input data sets.

Ensemble learning method is used for finding better performance and it integrates multiple learning algorithms and produces more performance than the individual algorithm. This type of learning has two main categories namely serialization and parallelization. Serialization refers to the existence of strong dependency between some individual learners that generate result serially and including boosting. Parallelization has no dependence with other learners and therefore the learners can be trained concurrently, including random forest and bagging [6].

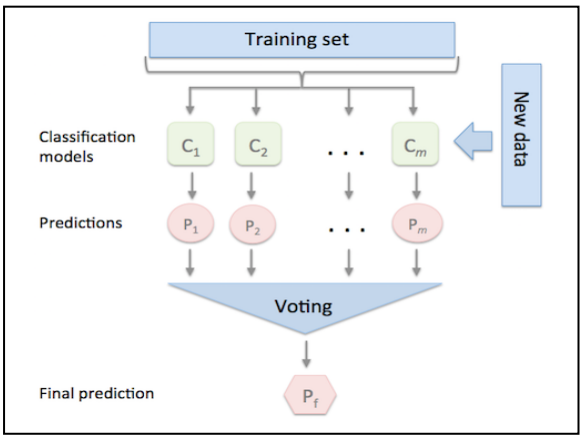

Figure 1. Voting for Accuracy

The basic idea of ensemble learning apply different learning models to get better classification or results [6]. The results from different classification models can be combined in two different ways, that are voting and averaging. Voting method is mainly used in classification and averaging method is commonly used in regression model. There are two common types of voting, hard and soft voting. Hard voting is also known as majority voting and in which each classifier votes individually and the majority of these votes is accepted. In soft voting, each classifier defines the probability values for a particular target class on each data point. By averaging these probabilities, the target label with the greatest average provides the vote [11].

Ensemble learning is also consolidation of some base models of machine learning methods to get one ideal model. Ensemble model can improve accuracy and robustness on single learning methods and also can overcome the constraints of a single method. Therefore it has some different learners called base learners. Base learners are known as powerless learners and their results are combined to get superior to strong learners [6]. Healthcare analytics is a supervised classification problem. Ensemble learning model proposed in this paper combines the results of three algorithms namely, Naïve Bayesian, Decision Tree and K-NN. The method for building ensemble learning and the algorithms used in ensemble learning are presented in the next subsection.

A. Ensemble Learning

Ensemble learning is a method that combines some machine learning algorithms to get better performance. The ensemble learning model is built in two steps. In the first step, all the base learners are used in parallel where the generation from a learner has an impact to the other learners. In the next step, the decisions or results of all base learners are combined in two different way namely, majority voting and weighted averaging. Result combination with majority voting is popular for classification and weighted averaging is popular for regression [6].

B. Machine Learning Algorithms Used

Ensemble learning is a method that combines some machine learning algorithms to get better performance. The ensemble learning model is built in two steps. In the first step, all the base learners are used in parallel where the generation from a learner has an impact to the other learners. In the next step, the decisions or results of all base learners are combined in two different way namely, majority voting and weighted averaging. Result combination with majority voting is popular for classification and weighted averaging is popular for regression [6].

i. Naïve Bayesian

Naïve Bayesian classifier is probabilistic classifier and that is based on Bayes theorem. This classifier is highly scalable and requires a number of parameters (features/predictors). Naïve Bayes is a simple method for building classification model. Class labels are assigned to problem instances and the class labels are drawn from some finite set. This classifiers can be trained in a supervised learning efficiently. It can be used in many complex situations in real-world environment. This method is outperformed by other approaches such as boosted trees or random forests. Naïve Bayesian classifier is used in application with automatic medical diagnosis. [9]. The advantage of Naïve Bayesian classifiers is that small number of training data is required to estimate for classification. The process of Bayes theorem is mathematical and to find the probability for a condition, that is mostly

related with a condition already taken. Bayes' theorem is based on the following equation:

$$P(A|B) = \frac{P(B|A)P(A)}{P(B)} \quad (1)$$

ii. Decision Tree

Decision tree is the form of tree structure and is used for classification or regression models. The data set is broken down into smaller and smaller subsets and therefore an associated decision tree is incrementally developed. Decision tree produces decision nodes and leaf nodes at its final result. Each decision node has two or more branches. The leaf node represents a decision or final result. The topmost decision node in a tree is root node. The root node is best predictor. A decision tree is also a top-down structure and the topmost is the root node. The data is partitioned into subsets that have similar values (homogenous). Entropy value is used in decision tree algorithm to get the homogeneity of a subset. The entropy is zero for the sample with completely homogeneous. If the sample is an equally divided, the entropy is one [5]. The entropy is calculated as:

$$H(x) = -\sum_{i=1}^{n} P(x_i) log_2 P(x_i) \quad (2)$$

iii. K-NN

K-NN or K-Nearest Neighbors is a simple algorithm and it can be used for both classification and regression. In both cases, the input contains the k closest training examples for the feature space and the output depends on whether k-NN is used for classification or regression. All available cases are stored and new cases is classified using a similarity measure or distance function. The case is assigned to class of its nearest neighbor when K is 1. Weights can be assigned to the contributions for the neighbors and therefore the neighbors nearer can contribute more on average than the more distinct ones. Let d is the distinct to neighbor, a common weighting scheme contains each neighbor a weight of 1/d to each neighbor. The neighbors are in the set of objects. For the neighbors, the class is for K-NN classification and the object property value is for K-NN regression.

C. Proposed Model of Ensemble Learning

This paper proposed an ensemble learning model based on soft voting method. Compared to hard voting method, the class labels can be predicted based on the predicted probabilities p for classifier. Firstly, healthcare data are analyzed using base learners. Big data analytics framework, Spark, and its MLlib (Machine Learning Library) are used for these base learners: Naïve Bayesian, Decision Tree and K-NN. Then accuracies from these base learners on each data object are obtained and soft voting is applied on the average of these accuracies. The result of soft voting is used as prediction value of ensemble learning. Let A has the probabilities, 0.9 for positive and 0.1 for negative on class label, B has 0.8 for positive and 0.2 for negative, and C has 0.4 for positive and 0.6 for negative. Then the soft voting method produces positive as its prediction as it has the average value 0.7 greater than those of negative. The prediction value for the soft voting is calculated as:

$$y = max_i(\sum_{j=1}^{n} P_{ij}/n) \quad (3)$$

IV. EXPERIMENTAL RESULTS AND ANALYSIS

In this experiment, three base learners or machine learning algorithms: Naïve Bayesian, Decision Tree and K-NN are used individually and their accuracy results are compared with those of proposed ensemble learning model. Experiments are done with Spark MLlib with Java. All techniques are implemented on a computer system with 8GB RAM, Intel Core i5 processor and Spark 2.4.4 framework on Mojave MacOS. The different datasets on healthcare are taken from UCI machine learning repository [12].

TABLE I. DATASET DESCRIPTION

Data set No.	Data sets Name	No. of Instances	No. of Attributes
1	Lung-cancer	598	57
2	Heart-disease	370	14
3	Diabetes-disease	100000	55
4	Cervical-cancer	858	36

TABLE II. ACCURACY COMPARISON

Data set No.	Classification Accuracy (%)			
	Naïve Bayes	Decision Tree	K-NN	Ensemble Learning
1	88.32	98.65	96.24	99.93
2	87.19	91.48	86.32	93.56
3	80.22	96.50	87.68	98.06
4	89.51	96.85	96.15	98.82

Each of base learners is trained with the training dataset and then testing datasets is applied to get the accuracy of each learner. For each of class label obtained by each data object with each base learner is recorded. According to soft voting method, these class label values are averaged to get the value for class label of ensemble learner. Then testing dataset is applied on each base learner and ensemble

learner. The accuracy of each base learner is recorded and are shown in table 2.

In this experiment, four healthcare datasets from UCI machine learning repository, namely Lung Cancer, Heart Disease, Diabetes and Cervical Cancer, are used. As shown in table 2, the accuracy of Naïve Bayesian classifier is minimum for all datasets whereas those accuracies of Decision Tree and K-NN classifiers fluctuate for all datasets. Nevertheless, proposed ensemble learning has the highest accuracy rate. The most attractive reason of using ensemble learning is to get more accuracy. This experience also shows that the ensemble method gets more accuracy than any single method

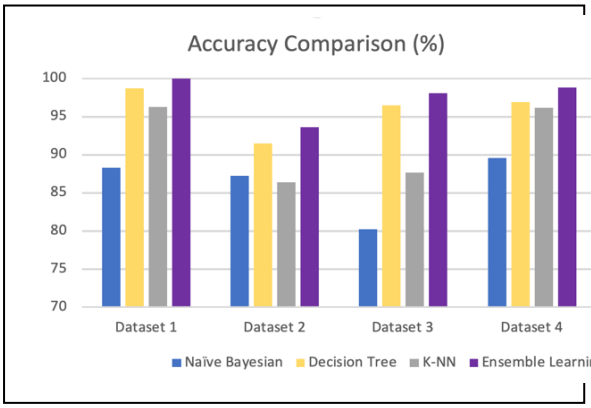

Figure 2. Accuracy Comparison

V. CONCLUSION AND FUTURE EXTENSION

This paper proposed an ensemble learning model that is based on the accuracy values of base learners: Naïve Bayesian, Decision Tree and K-NN classification algorithms. Soft voting method is used for combining accuracies of the base learners. By using this method, the proposed ensemble learning method has the highest accuracy than those of individual classifiers. There are many classification algorithms and many combining methods. Ensemble learning can also be applied with other classifiers and combining methods. Comparing the accuracies on these ensemble learnings are future works of the paper.

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Efficient Mapping for VM Allocation Scheme in Cloud Data Center

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Abstract

The allocation of virtual machines in the cloud data center has been a challenge in recent years. The needed resources for virtual machine (VMs) allocation map to the available resources of physical machines (PMs). The efficient allocation of the virtual machines is one of the optimization problems in order to resolve the underutilization of resources in cloud data center. The recently studies emphasized this optimization problem. This paper presents the proposed efficient mapping scheme of virtual machines (VMs) allocation in the cloud data center and then evaluates the efficiency of the proposed allocation scheme. In the presented mapping scheme, the VM allocation problem solved as Bin-packing with BestFit algorithm (BF) and the binary search concept is integrated. The efficiency of the proposed mapping scheme of VMs allocation is verified through the CloudSim simulator. According to the experiment results, the proposed binary search based VMs allocation requires less processing time than the existing one.

Keywords— *Bin-packing, BestFit, Binary Search, Cloud Data Center, Virtual Machine allocation, mapping scheme, CloudSim*

I. INTRODUCTION

The advent of cloud computing technology has turned the IT industry into a part of the modernized world, and large cloud data centers are serving the millions of on demand services for computing resources, storage, networks and applications based on a pay-per-use model. The virtualization technique enables the sharing of physical resources by allowing the number of virtual machines to be allocated to the sets of physical machines (PMs) in Data Center. The effectively utilize the resources of PMs become an important issue in IaaS. The proper VM allocation algorithms play as an important role in order to achieve a better utilization of PMs' resources in cloud data center. The Bin-packing model is an effective model to solve the virtual machine allocation problem. It enables to reduce the number of used PMs due to its packing techniques. There are recent studies [1, 2, 6, 7, 8] to resolve the VMs allocation problem in Data Center by applying Bin-packing model with FirstFit or BestFit scheme.

We proposed the resource management model and efficient VMs allocation algorithm in [3]. In which, the VM allocation algorithm applied the Bin-packing and BestFit Descending concepts to solve the utilization problem. The proposed algorithm considers not only the utilization of resources of PMS but also the reduction of mapping to suitable PMs for VMs allocation. In order to reduce the mapping and allocation time, the binary search mechanism is used in the proposed algorithm. Two resources- CPU and memory are used as configuration parameters for VMs allocation. The simulation result verifies that the proposed VMs allocation is an efficient mapping of VMs allocation.

The rest of the paper is organized as follows. Section II describes about the related work. Section III presents the proposed VMs allocation model and proposed VMs algorithm. Section IV explores the experiments and comparative analysis which verify our contribution. Section V conclude this paper and describes our future work.

II. RELATED WORKS

The efficient VMs allocation becomes a key factor and it is still a problem in cloud data center. The improper VMs allocation causes the underutilization of resources of PMs in data center. In [4], their proposed algorithm (RVMP) is able to scale down the active PMs in order to minimize the power consumption of IaaS cloud. In their study, the resource usage factor is contributed in their resource usage model in order to efficiently utilize the resources of the active PMs. In [1], Dynamic Resource Management Algorithm (DRMA) was proposed, and it can solve the resource utilization problem in a certain period of time, and the bin-packing and best fit method is used to solve resource allocation problem. For their simulation, two resources- CPU and RAM are used as configuration parameters. In [6], a virtual machine dynamic forecast scheduling (VM-DFS) is proposed in order to optimize the VMs allocation according to the forecast of future consumption in a cloud computing environment. VM-DFS analyzed historical memory consumption and predicted future memory consumption and allocated VMs to the most proper PM in the cloud datacenters. The problem of VMs allocation is formed as the

problem of bin-packing and used first-fit decreasing method. VM-DFS was based on dynamic deployment, time series forecasting theory, and binary packaging models. According to the experimental result, it can reduce the number of active PMs relatively. In [7], author explored two works- mapping of VMs to PMs and mapping of tasks to VMs and examined how these two works influence in each other. In [8], the author explored two heuristic algorithms–Modified BestFit Descending (MBFD) method's performance and contribution of Migration Minimization (MM) to consolidate VMs in a cloud data center. According to their study, these algorithms are practically useful for the issue of consolidating VMs. Our study focuses on modifications to existing models with an optimized VMs allocation algorithm based on binary search contribution [3] and verify the efficiency of mapping for VMs allocations by CloudSim simulator in this paper.

III. PROPOSED VIRTUAL MACHINES ALLOCATION MODEL

In cloud computing environment, Data Center is a specialized IT infrastructure that houses the number of physical machines. In our proposed model, Data center includes the number of physical machines (PMs) and that can host the number of virtual machines (VMs). There are two modules- VM manager and allocation manager in the cloud data center as shown in Fig.1. The VMs manager keeps tracks the resources of PMs in the cloud data center and the incoming VMs requests as well. Before allocation of VMs, the VM manager sorts the requested VMs and PMs in descending order according to their resources. The allocation manager allocates the VMs to the most proper PMs through the proposed Efficient VMs Allocation Algorithm. Here, we consider two resources-CPU and RAM for allocation problem. Initially, all PMs in the cloud data center are active.

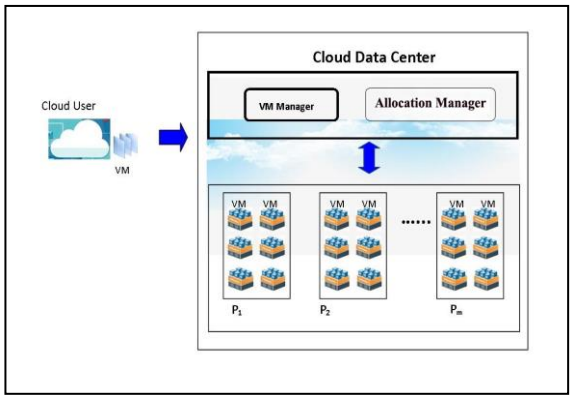

Figure 1. The proposed VMs allocation model

The CloudSim simulator enables to implement the cloud computing environment. The proposed VMs allocation model that describe in Fig. 1, is simulated by using the CloudSim 3.0. The CloudSim simulator provides DataCenter class, vm class, Cloudlet class. The DataCenter class is used for creating a Datacenter with a number of PMs and the vm class is used for creating virtual machines (VMs). The Cloudlet class is used for creating the tasks run on the virtual machines. In our proposed model, we consider the part of VMs allocation to suitable PMs in Data Center and not consider the part of task allocation to suitable VMs. To simulate the proposed model, the number of PMs and the number of VMs are created by DatacenterBroker class and then Datacenter class is created with those PMs. These VMs and PMs are keeping in sorted list before mapping and allocation. The BinaryVmAllocationPolicy class is created to maintain the Data Center allocation policy. The proposed model simulation handles with a SimulationStartingPoint main class.

A. Proposed VMs Allocation Algorithm

The efficient VMs allocation algorithm was proposed in our previous work [3].

Input: P, V, m, n
Output: P_{active}, VSetp, s, c=0
where P is PM list, V is VM list
P_{active} is currently active PM list
VSetp is current VM list in for each PM
m is number of Virtual Machine
n is number of Physical Machine
c is number of active PM
s is number of non-active PM
BEGIN
Sort P by CPU and Memory in descending order
Sort V by CPU and Memory in descending order
for each $p \in P$
VSetp ← Null ; // initially no running VM set for each PM
endfor
for each $v_i \in V$, $i \in \{1,2,..,m\}$
first← 1; last ← n;
do while (first≤ last) // mapping to the best fitted PM
mid←(first + last)/2;
if ($v_i^{mem} > p_{mid}^{mem}$ && $v_i^{CPU} > p_{mid}^{CPU}$) then
last← mid-1;
else if ($v_i^{mem} < p_{mid}^{mem}$ && $v_i^{CPU} < p_{mid}^{mem}$) then
first← mid+1;
endif
enddo
if ($v_i^{mem} \le p_{mid}^{mem}$ && $v_i^{CPU} \le p_{mid}^{mem}$) then $q \leftarrow p_{mid}$;
else $q \leftarrow p_{mid-1}$;
endif
$VSet_q \leftarrow VSet_q \cup \{v_i\}$
$q^{CPU} \leftarrow q^{CPU} - v_i^{CPU}$; $q^{mem} \leftarrow q^{mem} - v_i^{mem}$
if q is not belongs to P_{active} then
$P_{active} \leftarrow P_{active} \cup \{q\}$;
c++;
endif
endfor
s=m-c;
END

Figure 2. The proposed VMs allocation algorithm

In this proposed algorithm, we consider the resource utilization of PMs and mapping to suitable PMs for VMs allocation. It is important that, the PM needs to be host the more number of virtual machine for better resource utilization. Bin-packing model can handle the resource utilization issue. The proposed algorithm uses the Bin-packing with BestFit Descending method to handle resource utilization. Moreover, we consider the VMs allocation in linear fashion is time consuming task while mapping and allocating to the proper PMs. In this algorithm, the contribution of binary search based mapping can handle this time consuming issue and can reduce the allocation time efficiently.

In the proposed algorithm, the list of PMs and VMs need to be sorted in descending due to binary search method. The detail steps of the VMs allocation algorithm is described in Fig. 2.

IV. EVALUATION

The proposed VMs allocation algorithm is evaluated with complexity analysis and running time analysis. And then compare with the ordinary linear allocation algorithm.

A. Complexity Analysis

The Bin-packing heuristics with BestFit decreasing approach can be adapted to the proposed allocation problem. The complexity of algorithm is O(log n) and it is faster than the ordinary sequential based mapping. The allocation time would not too much increase even there are more PMs exist for mapping to allocate.

B. Run Time Analysis

The proposed model is evaluated with the average time taken parameters that is measured while mapping to find out the closest match PM with the needs of VMs' allocation. The experiment is carried out by deploying CloudSim 3.0 with Window 7 OS, RAM 4.0 GB, processor core i5 and Eclipse IDE Oxygen version. The proposed binary search based VMs allocation is simulated with seven experiments. We setup four types of VMs and PMs as shown in Table 1 and Table 2.

TABLE1. VMs Specification

Type	RAM (MB)	CPU (core)
Small	128	2
Medium	512	4
Large	512	8
Ex-Large	1024	16

TABLE 2. PMs Specification

Type	RAM (GB)	CPU (core)
Small	128	2
Medium	512	4
Large	512	8
Ex-Large	1024	16

TABLE 3. Experiment Setup of Four Types of PMs

#Experiment	# PMs	Small	Medium	Large	Ex-Large
1	5	1	1	1	2
2	10	2	2	2	4
3	20	4	4	4	8
4	50	12	12	12	14
5	70	17	17	17	19
6	100	25	25	25	25
7	150	35	35	35	45

For each experiment, we create four types of 20 VMs including 5 Ex-Large, 5 Large, 5 Medium, and 5 Small and four types of PMs 5, 10,20, 50, 70, 100 and 150 respectively. We setup four types of PMs for seven experiments are described in Table 3.The experiments do 20 times repeatedly in order to get the correctness of simulation result and find the average time taken of mapping for VMs allocations are recorded as shown in Table 4.

TABLE 4. Average Time Taken of Mapping the Requested VMs with PM

Number of Hosts (PMs)	Number of VMs	Average Time Taken (mili seconds)
5	20	146.3
10	20	155.15
20	20	158.33
50	20	160.82
70	20	167.6
100	20	188.8
150	20	191.9

According to the experimental results, the average time processing of the proposed binary search based mapping does not becomes too large even increasing the number of PMs. This shows that the proposed VMs allocation is more efficient in processing time while mapping to find out the most suitable PMs.

C. Comparative Analysis

In this paper, the experiments are done for VMs allocation algorithm based on ordinary sequential based mapping and the proposed binary search based mapping and then compare their average time taken of VMs allocation. For experiments for both linear and binary based allocation,

the setup configuration of VMs and PMs are the same with Table1,Table2 and Table 3. The average time taken of both VMs allocations are as shown in Table 5. According to the average time taken in mapping for VMs allocation, the proposed allocation algorithm can reduce in mapping time.

TABLE 5. Ordinary Sequential Mapping vs Binary Mapping

Number of PMs	Number of VMs	Average Time processing Sequential Based Mapping (mili seconds)	Average Time processing Proposed Binary search based Mapping (mili seconds)
5	20	146.9	146.3
10	20	157.17	155.15
20	20	160.34	158.33
50	20	162.85	160.82
70	20	173.6	167.6
100	20	195.8	188.8
150	20	201.9	191.9

As shown in Fig.3, the proposed binary search based mapping is faster than that of ordinary sequential one.

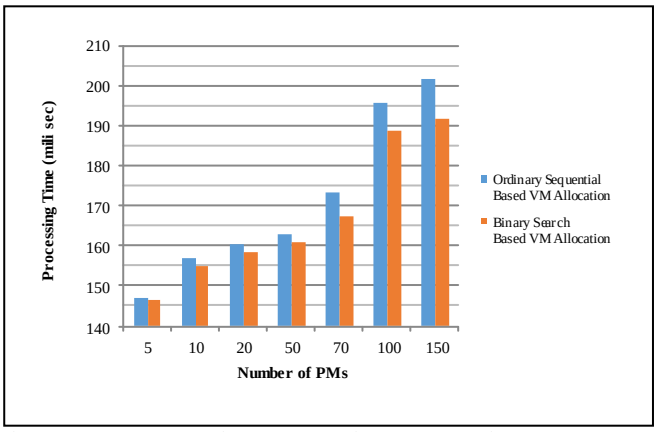

Figure 3. Average Time Processing Comparison

It is the contribution of the binary search during the mapping for VMs allocation. There is no doubt that the proposed VMs allocation is more efficient in mapping compare with the existing one.

V. CONCLUSION AND FUTURE WORK

In this paper, the efficient VMs allocation algorithm is presented and evaluates the efficiency of the proposed VMs allocation algorithm through CloudSim. For experiment of VMs allocation, two resources i.e memory and CPU are configured as resources' parameters. The comparative analysis shows that the average time taken of the proposed VMs allocation algorithm is faster than the existing one due to the contribution of binary search based mapping. In order to be an efficient the VMs allocation in Cloud Data center, the optimization problems- resource utilization and energy reduction are important. In the future, the effective policies and efficient algorithm would be integrated in the proposed VMs allocation model for resolving these optimization issues.

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Energy-Saving Resource Allocation in Cloud Data Centers

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Abstract

As the demands of cloud computing intensify, more facilities running power-intensive servers are necessary for cloud data centers. Such data center servers consume a large amount of energy, resulting in high operating costs. Therefore, high energy consumption has become a growing problem for cloud data centers. This paper proposes two energy-saving resource allocation algorithms that take into account several energy efficiency factors: resource allocation policies, power management techniques, and power models for better energy management in cloud data centers. These two algorithms are implemented on CloudSim toolkit and evaluated with real-world workload data. By the experimental evaluation of these algorithms for competitive analysis of energy consumption, we found that this work contributes to save the energy consumption in cloud data centers.

Keywords: *Cloud data centers, CloudSim, Energy consumption, Power management technique, Resource allocation*

1. Introduction

Cloud computing has become an essential aspect of the modern IT world, and cloud data centers become growing exponentially. Data centers' power consumption has increased in recent years due to an increase in size and number of data centers. The datacenter that delivers cloud services contains tens of thousands of servers. Data centers take advantage of virtualization technology [1] to host multiple virtual machines (VMs) on a single physical server. Electricity cost for powering servers forms a significant portion of the operational cost of data centers.

It is expected that the electricity demand for data centers to rise more than 66% over the period 2011–2035 [2]. It is estimated that energy costs may contribute even more than the cost of IT in the near future. Cloud service providers need to implement energy efficient management of data center resources to meet the increasing demand for cloud computing services and ensure low costs. Hence, there is a growing interest in saving energy consumption of cloud data centers.

Due to massive power consumption levels of data centers, energy-saving techniques have become essential to maintain both energy and cost-efficiency. Resource allocation is the most critical tasks in cloud computing. It involves identifying and allocating resources to every user request in such a manner that user requirements are met, and the goals of the cloud service providers are satisfied. These goals are primarily related to energy consumption or cost-saving.

Power management is also important since effective power management improves energy efficiency. As servers are the primary consumers of power in a data center, power management techniques are used to minimize power consumption by shutting temporarily down servers when they are not utilized. This research work also focuses on four energy-aware power models on cloud infrastructure environment.

CloudSim [3] is a generalized and extensible simulation toolkit that facilitates cloud infrastructures to be experimented and modeled. It supports for user-defined policies to allocate virtual machines to hosts. It is written in Java programming language and supports built-in classes to simulate the cloud environment on a single computing node. As a cloud environment consists of a large number of nodes, creating a real cloud and testing our proposed algorithms on it is impossible. So CloudSim offers a very suitable simulation environment to experiment our proposed energy-saving resource allocation algorithms.

This paper aims to manage data center power consumption and energy usage. In this paper, three different power management techniques and four energy-aware power models for two allocation policies are compared and analyzed to choose the most energy-efficient one. Based on these energy consumption comparisons, two energy-saving resource allocation algorithms are proposed and compared to gain higher energy efficiency. Cloud computing environments are simulated on the Cloudsim toolkit.

The remaining part of the paper is arranged as follows. Section 2 presents the literature review, section 3 discusses heuristics for energy efficient management, section 4 presents simulation setup, section 5 describes analysis for energy consumption of two allocation policies, section 6 presents proposed energy-saving resource allocation algorithms, and finally section 7 provides conclusion.

2. Literature Review

This section discusses state-of-the-art researches and technologies related to energy-saving resource allocation that eliminates a large portion of energy consumption in cloud data centers.

Ali et al. [4] proposed Energy Efficient VM allocation algorithm for data centers by selecting the most energy efficient host first. To reduce the power consumption in data centers, they applied three power management techniques: non power aware (NPA), power aware (PA) and dynamic voltage frequency scaling (DVFS) to their algorithms. Their algorithm achieved 23%, 23% and 9% more power efficiency than Best Resource selection (BRS), Round Robin (RR) and Minimum Power Difference (MPD) algorithms.

In [5], the researchers enhanced Round Robin (RR) algorithm by reintegrating Round Robin with shortest job first (SJF) algorithm that are selected for processing according to shortest task firstly in RR fashion then select the optimal job. This Scheduling algorithm gives a better result compared to RR.

Beloglazov et al. [6] proposed an energy-aware VM allocation algorithm that provision and schedule cloud data center resources to the user's tasks in an efficient manner. They demonstrated that their proposed algorithm reduces the level of data center's energy consumption.

An interior search based VM allocation algorithm: Energy-Efficient Interior Search (EE-IS) is proposed by the authors [7] for saving energy consumption and proper resource utilization. The proposed algorithm is implemented and tested on CloudSim and compared the amount of energy consumption with Genetic Algorithm (GA) and Best-fit Decreasing (BFD) algorithm. They showed that average 30% of energy has been saved using their proposed EE-IS as compare to the energy consumption of GA and BFD.

The research paper [8] evaluated the performance of four different power models: square root, linear, square and cubic models over IaaS Cloud infrastructure. They verified that cubic power model consumes less power than other three models.

This paper proposes two energy-saving resource allocation algorithms: DVFS enabled first come first serve (DFCFS) and DVFS enabled shortest job first (DSJF) considering energy efficiency factors: allocation policies, power management techniques and power models. We extend CloudSim to enable energy-saving resource allocation for data centers and to evaluate the performances of the proposed algorithms.

3. Heuristics for Energy Efficient Management

The total energy consumption is calculated by multiplying the power and time needed to turn on the servers. The energy efficiency objective aims to minimize the power consumption of servers in cloud data centers. Several energy-saving techniques can be used to monitor and control energy consumption.

3.1 Resource Allocation Policies

Resource allocation is the process of creating VM instances that match with the incoming requests onto hosts (servers). This paper emphasizes the following two allocation policies.

First Come First Serve (FCFS) – It is the simple allocation strategy. The request which comes first to the data center is allocated to the VM first. The only data required by allocator to make allocating decision is the arrival time of the request.

Shortest Job First (SJF) - The request is allocated to the VM with least run time among the requests in the ready queue. The request is always assigned to the process with least run time requirement. If the two requests having the same length, next request to be allocated, FCFS scheduling is used i.e. one which arrives first, will be allocated first to VM.

3.2 Power Management Techniques

Several techniques have been proposed for managing power consumption of data centers. The following are the existing techniques used for reducing power consumption without degrading the performance of servers in data centers:

Non Power Aware (NPA) - It calculates the server's energy usage without using any energy-saving method. Since there is no energy-saving mechanism, the total energy is dependent on the power consumption of the switched on servers and is independent of the CPU utilization. The servers use

the same amount of maximum power for both levels of extremely low and high CPU usage.

Power Aware (PA) - It calculates the energy consumption of servers independent of the CPU utilization as NPA and the utilized servers consume the same amount of maximum power for both extremes of too low and too high CPU utilization. It supports to shut down unused machines in the data center.

Dynamic Voltage and Frequency Scaling (DVFS) - It is able to save power consumption of a CMOS integrated circuit, a modern computer processor. The most of power consumption in CMOS circuits is composed of static and dynamic power. The dynamic power consumption is defined by multiplying the voltage square with system frequency as in (1). It scales the power of the system varying both CPU frequency and voltage. System voltage and frequency of a server can be adjusted by DVFS technology [9] without restart the power.

$$P_d = a * c * v^2 * f \quad (1)$$

where P_d is dynamic power consumption, a is switching activity, c is capacitance, v is voltage, and f is frequency.

Fan et al. [10] have found a strong relation between the total power consumption of a server and CPU utilization that power consumption of a server grows linearly with the growth of CPU utilization. DVFS is the dynamic power management technique [11] which reduces the dynamic power consumed by dynamically changing the frequency and the voltage of the processor during execution depending on the CPU utilization. By the time CPU voltage is decreased depending on the CPU utilization, a huge amount of energy is saved. Therefore, the cloud service providers can increase their profit by reducing the dynamic power consumed.

3.3 Power Models

By using the utilization of the CPU server and its power consumption in idle and maximum states, the power consumption of CPU servers can be estimated by power models. CloudSim provides an abstract "Power Model" implementation that can be extended to support various power models [12]. In recent releases of CloudSim, the provided power models are as follow:

Linear model: $P(u) = P_{idle} + (P_{max} - P_{idle}) * u \quad (2)$

Square model: $P(u) = P_{idle} + (P_{max} - P_{idle}) * u^2 \quad (3)$

Cubic model: $P(u) = P_{idle} + (P_{max} - P_{idle}) * u^3 \quad (4)$

Square root model: $P(u) = P_{idle} + (P_{max} - P_{idle}) * \sqrt{u} \quad (5)$

where current CPU utilization u, the maximum power value P_{max} and the idle power value P_{idle} of the CPU server.

4. Simulation Setup

CloudSim toolkit [3] is used to simulate the virtualized cloud computing environment. We extend CloudSim to allow energy-saving resource allocation for cloud data centers.

The real-world workload traces chosen for the experimentation is the RICC dataset (RIKEN Integrated Cluster of Clusters), publicly available at Parallel Workload Archive [13]. There are enormous collections of workload traces (dataset) of a variety of High Performance Computing. It contains trace of several thousands of submitted job requests over a period of five months; each has arrival time, run time (length), amount of requested CPU and memory.

To evaluate the proposed algorithms, we consider a cloud infrastructure provider with 3 data centers and, each data center has 45 heterogeneous physical servers with five different configurations shown in Table 1.

Table I. Server types characteristics

Server Type	Number of Core	Power	Number of Servers
Type 1	64	1100 W	9
Type 2	64	750 W	9
Type 3	32	750 W	9
Type 4	28	800 W	9
Type 5	16	750 W	9

In this work, we modify CloudSim to enable the simulation of High Performance Computing jobs from the logs of workloads on parallel machines [13]. At the start of simulation, VMs are created, and jobs (cloudlets) are submitted to data center broker which maps a job to a VM. Since we focus on allocation of VMs to physical hosts, we create a one-to-one mapping between cloudlets and VMs. Moreover, we implement VM termination during simulation to ensure complete simulation. The allocation of VMs to hosts utilizes our proposed algorithms. Space shared policy is used to assign the cloudlets to VMs, so that the jobs are sequentially executed in each VM. Using this policy, each job unit has its own dedicated core, and thus number of incoming jobs or queue size did not affect the execution time of individual job units since the proposed algorithms use non-preemptive method. Power management techniques and power models are implemented using the class PowerModel. This class offers getPower() function, that, returns the

power consumption of hosts depending on power management techniques and power models. CloudSim's inner code is modified to evaluate our proposed algorithms and to compare them. Then our own allocation classes are specified to extend the basic CloudSim classes.

5. Analysis for Energy Consumption of Two Allocation Policies

Three power management techniques: NPA, PA and DVFS, and four energy-aware power models: square root, linear, square and cubic are applied in two allocation policies: FCFS and SJF. Energy consumptions are compared and analyzed in each allocation policy to choose the most efficient one for better energy management in cloud data centers.

5.1 Energy Consumption Comparison in FCFS Resource Allocation

The comparison of energy consumption with NPA, PA and DVFS techniques in FCFS allocation policy for different number of requests are shown in Figure 1. DVFS has lower power consumption in comparison with PA and NPA because these two mechanisms use maximum power for all servers although PA supports to shut down the servers which are not used and NPA does not support to shut down the unused servers. DVFS consumes power depending on the CPU utilization of the utilized servers.

Table 2. provides energy consumption (kWh)

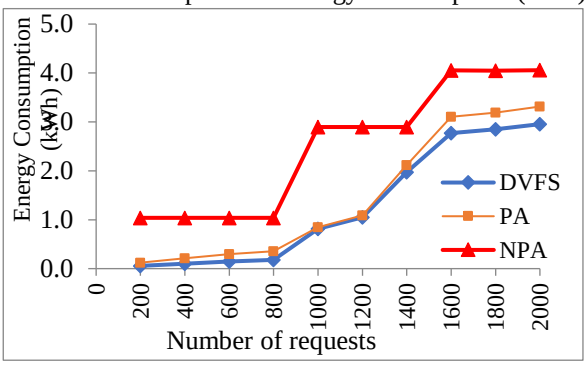

Figure 1. Comparison of energy consumption with different power management techniques in FCFS resource allocation policy

with various power models of DVFS in FCFS allocation policy for the different number of requests. It can be seen that the cubic power model is the most effective one compared to the other three models for every number of requests. This model can, therefore, be used to save energy in data centers.

Table II. Energy consumption comparison of DVFS with four power models in FCFS resource allocation policy

Number of requests	Square root	Linear	Square	Cubic
200	0.05716	0.05716	0.05666	**0.05659**
400	0.10476	0.10361	0.10275	**0.10254**
600	0.14685	0.14611	0.14538	**0.14510**
800	0.17843	0.17778	0.17700	**0.17661**
1000	0.81771	0.81339	0.80895	**0.80704**
1200	1.05271	1.04889	1.04533	**1.04371**
1400	1.97530	1.97027	1.96196	**1.95536**
1600	2.77189	2.77059	2.77025	**2.77023**
1800	2.84846	2.84661	2.84552	**2.84533**
2000	2.95514	2.95378	2.95194	**2.95084**

5.2 Energy Consumption Comparison in SJF Resource Allocation

The comparison of energy consumption with NPA, PA and DVFS techniques in SJF allocation policy for different number of requests are presented in Figure 2. It can be clearly seen that DVFS consumes less power than other two techniques NPA and PA because these two techniques use maximum power for all servers when the servers are on while DVFS consumes power depending on the CPU utilization, and shuts down the servers that are not used. Thus, a huge amount of energy is saved with DVFS.

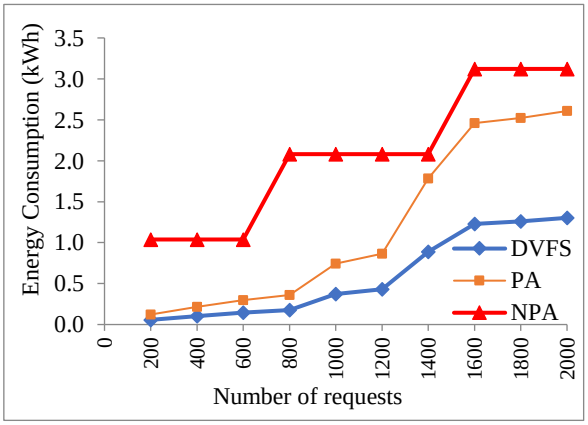

Figure 2. Energy consumption comparison of different power management techniques in SJF resource allocation policy

Table 3. presents energy consumption (kWh) with different power models of DVFS in SJF allocation policy for different number of requests. It can be found that energy consumption with the cubic power model is less than the other three models. Hence this model is applied in the proposed algorithms for better energy efficiency in the cloud data centers.

Table III. Energy consumption comparison of DVFS with four power models in SJF resource allocation policy

Number of requests	Square root	Linear	Square	Cubic
200	0.05821	0.05716	0.05666	**0.05659**
400	0.10476	0.10361	0.10275	**0.10254**
600	0.14685	0.14611	0.14538	**0.14510**
800	0.17843	0.17778	0.17700	**0.17661**
1000	0.37145	0.37133	0.37110	**0.37090**
1200	0.43174	0.43095	0.42990	**0.42927**
1400	0.89015	0.88819	0.88470	**0.88168**
1600	1.22865	1.22711	1.22519	**1.22418**
1800	1.26036	1.25903	1.25724	**1.25616**
2000	1.30356	1.30326	1.30276	**1.30235**

6. Proposed Energy-Saving Resource Allocation Algorithms

In this paper, two energy-saving resource allocation algorithms: DFCFS and DSJF algorithms are proposed based on the results of the above experiments. The energy consumptions of these two algorithms are compared to choose the better energy efficient one.

6. 1 Proposed DFCFS Algorithm

Energy-saving resource allocation DFCFS algorithm shown in Figurer 3. is proposed based on FCFS scheduling tasks and DVFS power management technique with cubic power model.

Algorithm1: DFCFS Algorithm

1. *procedure Resource Allocation (VM_j, $Host_i$)*
2. *Add user request to VM based on matching configuration*
3. *for all VM from j to n do*
4. *for all Host from i to m do*
5. *if VM_j fits in $Host_i$ then*
6. *Calculate the remaining CPU capacity of Host after VM has been added*
7. *end if*
8. *end for*
9. *Start a new Host in the data center, and allocate remaining VM into a new Host*
10. *end for*
11. *Calculate total energy consumption of servers with DVFS and cubic power model*
12. *end procedure*

Figure 3. DFCFS Algorithm

6.2 Proposed DSJF Algorithm

In DSJF algorithm shown in Figure 4, the resource allocation policy SJF is applied and the energy consumption of the servers are calculated by using DVFS with cubic power model chosen according to the above comparative and analysis results.

Algorithm2: DSJF Algorithm

1. *procedure Resource Allocation (VM_j, $Host_i$)*
2. *Sort the request list in ascending order according to their length*
3. *for all requests in the sorted List do*
4. *Add user request to VM based on matching configuration*
5. *if the current request length = next request length then*
6. *Sort the request according to their arrival time*
7. *end if*
8. *end for*
9. *for all VM from j to n do*
10. *for all Host from i to m do*
11. *if VM_j fits in $Host_i$ then*
12. *Calculate the remaining CPU capacity of Host after VM has been added*
13. *endif*
14. *end for*
15. *Start a new Host in the data center, and allocate remaining VM into a new Host*
16. *end for*
17. *Calculate total energy consumption of servers with DVFS and cubic power model*
18. *end procedure*

Figure 4. DSJF Algorithm

6.3 Energy Consumption Comparison of DFCFS and DSJF

Figure 5. shows comparison of energy consumption with DFCFS and DSJF algorithms for different number of requests. The energy consumptions for DFCFS and DSJF from 200 to 800 numbers of requests are almost indistinguishable. The reason behind is that the run time of those requests are similar in spite of sorting ascending order as DSJF algorithm. As the algorithms mentioned above, DSJF sorts ascending order for the run time (length) of requests before execution, while DFCFS does not. There is no impact of DSJF algorithm for the requests having almost equal run time. When the requests over 800 having different run time, it can be seen the impact

of DSJF algorithm that can save energy consumption compared to DFCFS algorithm. DSJF takes minimum turnaround time because the shortest length task gets finish in shortest possible time. Then the VM that has now become idle after executing the current task can take up next selected task. This will minimize the number of active VMs as well as active servers so that power consumption is less. In this way, DSJF resource allocation can greatly reduce power consumption. DSJF algorithm gains higher energy efficiency for the case where run time of incoming requests are different from each other and it can save up to 55% of energy consumption compared to DFCFS algorithm.

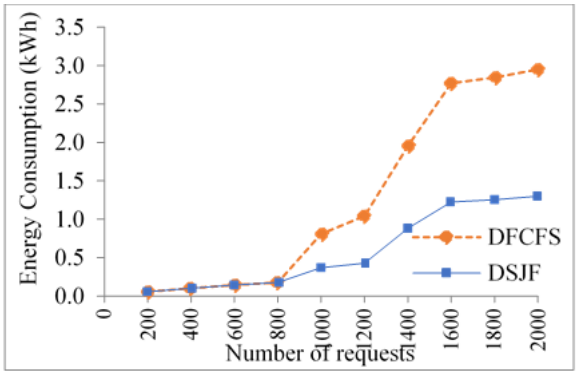

Figure 5. Energy consumption comparison of DFCFS and DSJF

7. Conclusion

Cloud data centers are the digital age factories, and data center power consumption becomes a global issue. The proposed algorithms are intended to save cloud data center energy consumption. To achieve this goal, under FCFS and SJF allocation policies, different power management techniques and different power models are compared and analyzed. The results proved that DVFS is saving more power than the other two power management techniques. The findings of the analysis showed that the cubic power model achieves greater energy efficiency and consumes less power than the other three power models. The evaluations for the proposed algorithms: DFCFS and DSJF are experimented using real workload traces of virtualized environment on Cloudsim simulator. The combination of SJF resource allocation algorithm and DVFS power management techniques with Cubic power model, called DSJF is very suitable for the case where run times of incoming requests are different from each other. The experimental results show DSJF algorithm gains higher energy efficiency and it can save up to 55% of energy consumption compared to DFCFS algorithm.

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Ensemble Framework for Big Data Stream Mining

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Abstract

The rapid development of industry enterprises, the large amount of data generated by these originalities and the exponential growth of industrial business website are the causes that lead to different types of big data and data stream problem. There are many stream data mining algorithms for classification and clustering with their specific properties and significance key features. Ensemble classifiers help to improve the best predictive performance results among these up-to-date algorithms. In ensemble methods, different kinds of classifiers and clusters are trained rather than training single classifier. Their prediction machine learning results are combined to a voting schedule. This paper presented a framework for stream data mining by taking the benefits of assembling technology based on miss classification stream data. Experiments are carried out with real world data streams. The experimental performance results are compared with the modern popular ensemble techniques such as Boosting and Bagging. The increasing in accuracy rate and the reducing in classification time can be seen from the test results.

Keywords: *Big Data, Bagging, Boosting, Data Stream Mining, Ensemble Classifiers, Misclassification Stream Data*

I. INTRODUCTION

In the strong-growing of big data era, all the internet application significantly needs to process large amount and varieties of data. This growing is quickly rapid up and affecting to all technology and businesses environments for organizations and individuals benefits respectively. Furthermore, big data analysis intend to extract the statistical information using data mining algorithms in instantaneously way that assist in making likelihoods, finding the hidden information, classifying recent developments and defining decisions. Though, the rise in classification speed comes at what cost, difference in estimation with the original, and mis-assigning in relative classes whatever machine learning algorithms used [1-3].

To overcome this problem, this paper focuses on finding a way to speed up the mining of streaming in high accuracy rate based on miss classification data streams.

Section 2 deals with the nature of big data and how it is associated in real-world applications. Section 3 describes about the data stream mining, and briefly describes modern data stream mining methods. Section 4 provides the previous research for data stream classification. Section 5 proposes the classification framework. Experimental setups and results are depicted in Section 6. Finally, conclusions are presented in section 7 to summarize outcomes.

II. BIG DATA

The meaning of "Big Data" can be classified into many ways: someone defines that big data is the large amount of data over a certain threshold. Others defined as data that cannot handle by the conventional analytical suits such as Microsoft Excel. More popular mechanisms identified big data as data that has the Variety, Velocity, and Volume features. Big data analytics is an innovative approach including of various mechanisms and procedures to extract treasured insights from raw data that does not suitable for the traditional database system due to any reasons.

Big data applications can be found in several fields such as financial area, technological area, electronic governmental area, business and health care processes, etc. Furthermore, in other specific cases, energy control used big data, anomaly detection, crime prediction, and risk management. Big data are having a strong control for every kinds of business.

Information data can be defined as a new form of investment, a different type of currency, and an original resource of valuable things. It has been revealed about the power of big data that has the efficient strategies to become successful business. But it can't be argued that all the strategies of big data may not be used for all business types. However, it is the universal truth that a data information strategy is still valuable, whatever the size of data. This enormous

amount of data in application opens new challenging detection tasks and lead to Data Stream Mining [4].

III. DATA STREAM MINING

In computer science, data stream mining is associated with two fields: data mining and data streams. It turns out to be essential areas of computer science applications such as industrial engineering processes, transaction flows of credit card, robotics, e-commerce business, spam filtering, sensor networks, etc.

The data stream mining task quite different traditional data mining task regarding processing or executing the mining task, but the objectives are the same. The normal algorithms of data mining methods cannot be used directly for data streams because of the following factors:

-Data Streams may be large amount of data and these are actually unlimited number of elements.

-Data Streams can be arrived to the system in a short period of time.

-Data Streams may be changed to different manners during the distribution times of processing.

Therefore, the algorithms for data stream need to store previous information in a condensed format structures. The most widespread methods for the data stream classification are categorized into the following groups;

- Instance-Based Learning Methods
- Bayesian Learning Methods
- Artificial Neural Networks Learning Methods
- Decision Trees Learning Methods
- Ensemble Learning Methods
- Clustering Methods

A. Instance-Based Learning Methods

Instance-based learning classifiers are also called k-nearest neighbors learners. As these instance classifiers can process incremental learning method, there is a necessary to store all the previous data elements in the memory. Therefore, these normal learning methods can't be directly used for data streams [5]. All the series of instance-based classification algorithms were presented in [7].

B. Bayesian Learning Methods

Bayesian learning classification methods are based on the standard Bayesian theorem. The aim of Bayesian learning is to evaluate the essential likelihoods using the existing training dataset. Then, a learning algorithm is used to categorize new data—the group which maximizes the next probability is allocated to an uncategorized or unlabeled element. Naive Bayes learning method is done in an incremental fashion. However, they need to have a fix size of memory. These Naive Bayes learning features possibly appropriate in the mining process of data stream [5].

C. Artificial Neural Networks Learning Methods

Artificial Neural Networks learning methods are likely the nervous system of the animals. Multi-Layer Perceptron is the most common learning classification method. When the number of data streams training elements is large, neural network learning can be transformed to single-pass incremental way. If input neurons and synapses number is kept unaltered during the learning process then the memory requirement is kept constant. The above properties of neural networks can be appropriate for data streams [5, 8].

D. Decision Trees Learning Methods

The state-of-the-art Decision Tree algorithms can be used for classification of data stream. The algorithms in this type are based on Hoeffding trees method. For the static data, Hoeffding tree chooses an attribute that is appropriate to split tree nodes. Because of the infinite size of the data streams, all the data elements of the node can't be kept in memory. Therefore, evolutionary learning algorithms are used for data streams. The most noticeable method of this type of learning is the VFDT algorithm [10].

E. Ensemble Learning Methods

Other learning techniques which can be applied for data stream mining are ensemble ways. Various approaches are suggested to combine many single algorithms into a group to form ensemble classifiers. Among the state-of-the-art classifiers available in many data mining environment, including the stream data mining, assembling of classifiers provides the best performance [5, 11-13]. The most common and effective approaches of ensemble methodologies are Bagging and Boosting. These popular methods have been discovered in the data stream scenario and are firstly presented by Oza and Russell [14, 15].

F. Clustering Methods

Clustering can be used for the unsigned instances that have homogeneous clusters relating with

their similarities. Streaming methods for clustering can be done with two levels, online and to eliminate similar data which in offline. At the online level, a set of extremely small clusters is computed and updated from the stream efficiently; in the offline phase, a classical batch clustering method, for example, k-means is performed on the micro clusters. Although the offline level clustering carries out for several amount of processing phase, the online level clustering only done with a single pass phase for the input data. Because the offline processing can be separated to a set of small clusters and can be invoked when the stream ends. Furthermore, they can update the group of separated clusters periodically according to the stream flows they need.

The k -means clustering method is one of the most used methods in clustering, due to its simplicity. To initiate the clustering process, the value of k is chosen in a random way, but most popular developed algorithms begins with 1, and some starts with 5 or 10. Then, according to that centroid value, each instance is assigned to the nearest centroid. The cluster centroids are computed again with the center of mass of the assigned instance. This process is computed repeatedly until the desired criterion is encountered or the assignments cannot be changed. This routine cannot be used for data streams mining process because the streams require many passes to be clustered [16].

However, this paper focuses on Bayesian Classifiers and Ensemble Classifiers and Decision Trees and Cluster.

IV. Literature Review

Hundreds of academic papers have been presented based on research done for big data classification on the standard dataset of data stream mining. These works done are described according to categorization of the above state-of-the-art classification groups.

The author in paper [6] illustrates the developed numerous streaming data computation platforms and discuss their major abilities. They clearly specify the prospective research directions for high-speed large-scale mining methods for data stream from different point of views such as procedures, implementation nature and performance evaluation analysis. They clearly described that Instance-Based Classifiers get more accuracy among the other classifiers but the time taken for that is extremely large. Therefore, in order to perform faster changes, the authors in [17] took the distributed computing advantages and then proposed the nearest neighbor incremental classifier.

In [18], an operational pattern-based Bayesian learning classifier was proposed to handle data streams. The researchers in [19] implemented highly efficient and popular algorithm “Naïve Bayes Algorithm” on huge complex data to acquire knowledge. They proposed reduction technique to remove similar data which in sequence reduces the time of computation, the amount of memory space requirement, and enhances the performance of Naïve Bayes Algorithm. Their research work indicates highly efficient Naive Bayes Algorithm solution or huge data streams.

Many authors proposed to apply Neural Network Deep Learning methods for data stream processing in many ways. Neural Network deep learning architectures can be capable for complex tasks and sometimes it can outperform human’s beings in some application areas. Although the remarkable advancements for this area can be seen clearly, there is an ill-posed optimization problem for training deep architectures that has a very large number of hyper-parameters. For this reason, modification of Neural Network integrated framework has been presented to get online calculation capabilities of highly scalable solution for mining data streams in [20].

Extensive surveys research on assembling for classification of data stream and also regression tasks have been done by [13]. They surveyed wide ranges of data streams ensemble techniques and introduces the innovated learning methods for imbalance data streams processing including complex representation of data, semi-supervised learning, the structured outputs and the detection method. The authors in [16] propose a new ensemble learning method, called Iterative Boosting Streaming ensemble (IBS) that can be able to classify streaming data. The authors in [21] introduce the new distributed training model for ensemble classifiers to avoid the dangers of the vote-based separated ensembles. This model is named as “LADEL”.

V. Proposed Framework

Let a data stream input as a sequence of batches DS = {S1, S2 . . ., St,}, where St = {S1…, SN} be an unlabeled batch. Assume the real class label Li of instance Si, for i = 1. . . N, and the equivalent labeled set, defined as ˆSt = {(S1, L1) . . ., (SN, LN)}, that can be used at the training stages. The class labels are necessary to be predicted manually for incoming unlabeled data of real-world data. The standard stream dataset no need to do this state. An automatic mining system of data streams, that has acceptable and

constant performance at classification accuracy, computation procedure and memory usage.

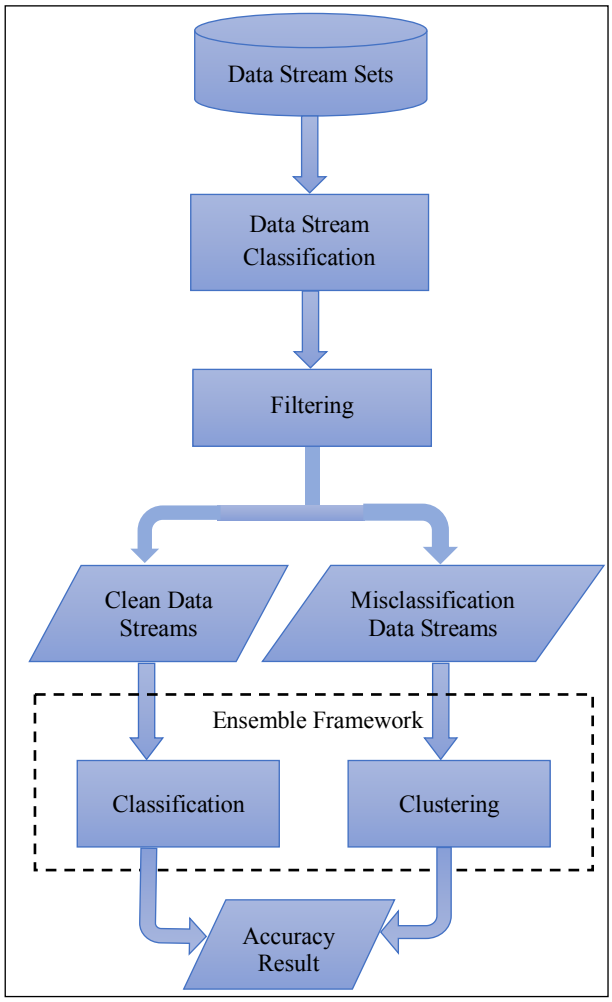

Figure 1. Ensemble Framework for Data Streams

After labeling batches, St, is presented, the simple classifier categorizes its instances. There is an assumption that the name of the labels will be known immediately after the classification process took place, so the miss classification error of this batch can be predicted and clearly know the correct instance stream. The uncertain, ambiguous, incomplete, and subjective data can reduce the performance of the classifier and not all the techniques are suitable for all data streams. Therefore, the misclassification data streams are filtering out after the classification process. After the filtering process is created, assembling method is designed by using the simple cluster to improve the model to focus on data stream that is not easy to classify. The miss classified data streams, MDS = {S1, S2… St} are separated from correctly labeled streams, clean data streams CDS= {S1, S2…, St} from DS. Then the clustering is ensemble to label the left incorrect data streams, MDS. After grouping batches of MDS, the accuracy for that clusters is calculated, and the overall accuracy of the DS can also be calculated as illustrated in Fig. 1.

VI. EXPERIMENT RESULTS

The well-known data set of streams, real world Electricity [22] data is used to test the ensemble effect. The Electricity data was accumulated from the New South Wales' electricity market, Australian State. Prices are unstable and depends on the demand and supply in this market. It contains the real data collected at every 30 minutes for 2 years and 7 months. This dataset consists of 45,312 instances with five attributes for the time, day, period and price. The class label defines the alteration of the price corresponding to moving average (MA) of the past 24 hours.

The experiments are conducted on the tasks of data stream classification. The experiment design is implemented with the use of EvaluatePrequential approach because the system needs to know the label of the class and to filter out the mis-classified data. Firstly, the classification is done with the well-known classifier Naïve Bayes and VFDT and the results are shown in Table I.

TABLE I. PERFORMANCE MEASUREMENT COMPARISON FOR SINGLE STANDARD DATA STREAM CLASSIFIERS

Name of Data Stream Algorithm	Classification Accuracy (%)	Kappa Statistic	Kappa Temporal Statistic	Elapsed Time (s)
Naive Bayes	73.07	40.89	-83.57	0.67
VFDT	72.23	43.59	-89.30	0.52

The data streams are tested with the standard ensemble methods of Leveraging and Boosting. The results from the experiments are summarized in Table II.

TABLE II. PERFORMANCE MEASUREMENT COMPARISON FOR ENSEMBLE DATA STREAM CLASSIFIERS

Name of Data Stream Algorithms	Classification Accuracy (%)	Kappa Statistic	Kappa Temporal Statistic	Elapsed Time (s)
LeveragingNB	52.82	12.95	-221.62	1.58
LeveragingVFDT	75.497	48.38	-67.04	4.25
OZOBoostNB	74.322	44.38	-75.04	1.02
OZOBOOSTVFDT	69.352	39.70	-108.92	1.72

Then experiments are carried out for the part of the proposed framework by using the simple KMeans clustering method. The data and results are shown in Table III and this addition tasks can enforce the

classification accuracy but need to care for elapsed time. The overall performance for the proposed ensemble method is illustrated in Table IV.

TABLE III. PERFORMANCE MEASUREMENT COMPARISON FOR CLUSTERING FOR MISS DATA

Data Stream	Classification Accuracy (%)	Data Count	Elapsed Time (s)
Miss data for NB	45.57	12202	0.03
Miss data for VFDT	59.29	12583	0.08

TABLE IV. PERFORMANCE MEASUREMENT COMPARISON FOR PROPOSED ENSEMBLE DATA STREAM CLASSIFIERS

Name of Data Stream Algorithms	Classification Accuracy (%)	Elapsed Time (s)	Improved Accuracy (%)
NB + KMeans	88.04	0.7	14.97
VFDT + KMeans	88.69	0.6	16.46

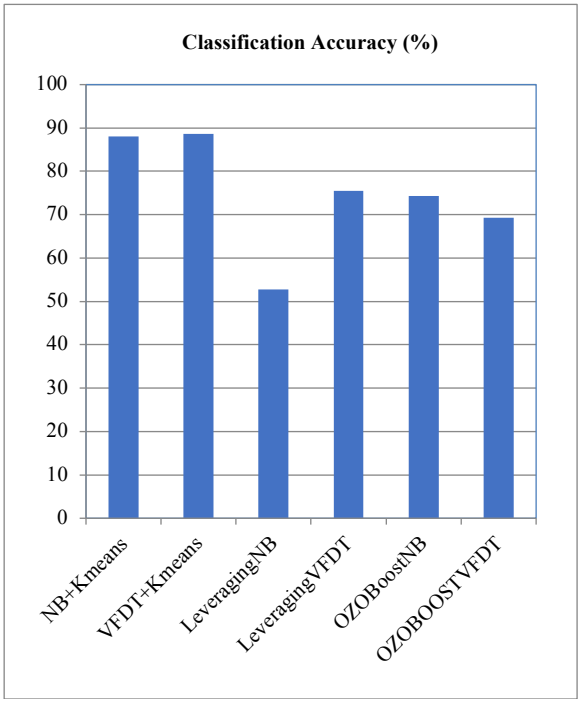

Figure 2. Measurement Comparison for proposed Ensemble Data Stream Classifiers and Standard Ensemble Classifier

Then the comparison is carried out the proposed ensemble method and the state-of-the-art ensemble methods and results are shown in Fig. 2 and Fig. 3. From these results, it can be seen clearly that the proposed framework not only can increase the classification accuracy but also less than in elapsed time.

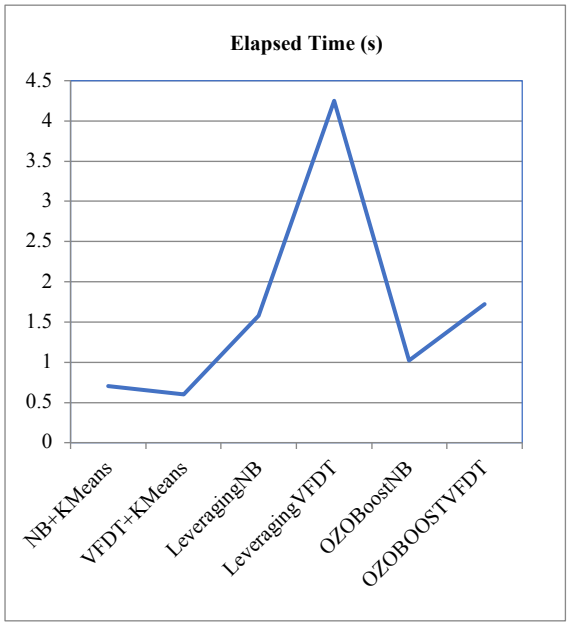

Figure 3. Time Consume comparison for proposed Ensemble Data Stream Classifiers and Standard Ensemble Classifier

VII. CONCLUSION

In this paper, the ensemble framework constructed from the data streams classifiers and simple K-Means clustering is proposed for mining data streams. The proposed framework of the ensemble learning classifiers, the combination of Naïve Bayes and K-Means, and VFDT and K-Means, has been evaluated. Furthermore, the comparison of the proposed framework against state-of-the-art ensembles, Leveraging and Boosting using standard data stream set. The results clearly show that the proposed framework not only can improve the classification accuracy based on mis-classification data, but also can reduce the time taken than the above standard ensemble techniques. Future research will concentrate on learning the influence of the size of stream data and more effective ensemble mechanisms on accuracy of the ensemble classifier.

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Improving the Performance of Hadoop MapReduce Applications via Optimization of Concurrent Containers per Node

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Abstract

Apache Hadoop is a distributed platform for storing, processing and analyzing of big data on commodity machines. Hadoop has tunable parameters and they affect the performance of MapReduce applications significantly. In order to improve the performance, tuning the Hadoop configuration parameters is an effective approach. Performance optimization is usually based on memory utilization, disk I/O rate, CPU utilization and network traffic. In this paper, the effect of MapReduce performance is experimented and analyzed by varying the number of concurrent containers (cc) per machine on yarn-based pseudo-distributed mode. In this experiment, we also measure the impact of performance by using different suitable Hadoop Distributed File System (HDFS) block size. From our experiment, we found that tuning cc per node improve performance compared to default parameter setting. We also observed the further performance improvement via optimizing cc along with different HDFS block size.

Keywords: *MapReduce, parameter tuning, concurrent containers, block size*

I. INTRODUCTION

Apache Hadoop is an open-source distributed software framework for large scale data storage and processing of data-sets on clusters of commodity hardware. Apache Hadoop architecture consists of the three main components: YARN (Yet Another Resource Negotiator), HDFS (Hadoop Distributed File System) and MapReduce. HDFS is used as a big data storage layer (for structured and unstructured data) in a distributed fashion with high throughput access to application data in a reliable manner and Hadoop MapReduce is the software layer implementing the MapReduce paradigm (only one of many possible framework which runs on top of YARN) and provides YARN based parallel processing of large data sets [5, 6]. YARN is an improved architecture of Hadoop that separates resource management from application logic [1]. The generalization of resource management makes it easier to deploy not only MapReduce applications, but also other applications such as Spark and Tez [1].

Despite the improvement in architecture of Hadoop in YARN and its wide adoption, performance optimization for MapReduce applications remains challenge because it uses static resource management based on pre-defined resource units (a unit could represent to 1GB memory and 1 CPU core) called containers that are assigned to the map tasks and reduce tasks of MapReduce application. Therefore, the size of a unit and the total number of available containers per node are static in nature, i.e., we need to determine it before creating the cluster and/or prior to executing the application and it cannot be changed after start up the cluster [4, 11]. This form of static resource management has a limited ability to address with diverse MapReduce application resulting in poor performance. Therefore, the performance limitation of Hadoop yarn framework are tested and measured by running representative MapReduce applications on pseudo-distributed mode with YARN. In MapReduce framework, the reduce stage depend on output from the map tasks and they can start as soon as any map task finishes. Therefore, map tasks are important to finish as quickly as possible. In this work, map tasks are selected to optimize the map elapsed time of MapReduce applications along with optimal cc. In other word, finding optimal cc per node is determining the optimal number of concurrent map tasks per node. The experimental result shows that optimal cc yields significant performance improvement over Hadoop-default configurations. The number of map tasks for a job depends on HDFS block size. For example, if the input file size is 1024MB, the number of map tasks is 8 and 4 for HDFS block sizes with 128MB and 256MB, respectively. Where, additional time of map tasks (such as creating/destroying JVM, setting up/cleaning up at task level, etc.,) can be reduced by reducing the number of map tasks, Therefore, we also analyzed the impact of performance by running on different suitable HDFS block sizes. Optimizing cc along with different HDFS

block size can achieve the further performance improvement over the default parameter setting.

In the following sections of the paper, we present background theory, determining the number of cc and why cc is selected for performance tuning, describe the performance evaluation on pseudo-distributed mode with Hadoop-2.7.2 and finally conclude the proposed system.

II. BACKGROUND THEORY

A. Hadoop MapReduce Framework

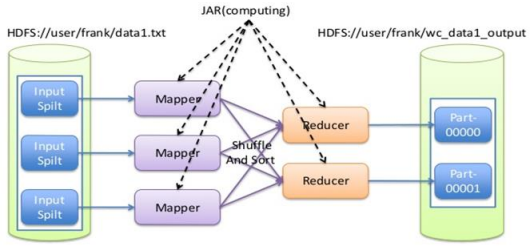

Figure 1. Overview of MapReduce Architecture

In Hadoop, the input files reside in HDFS (e.g., HDFS://user/frank/data1.txt) by dividing it into blocks with block size of 128MB. Hadoop InputFormat divides the input data into logical input splits in Hadoop. They just refer to the data which is stored as blocks in HDFS. In map stage, Hadoop creates one map task per input split and process records in that input split [13]. That is how parallelism is achieved in Hadoop framework. When a MapReduce application (JAR) is run, mappers start producing intermediate output internally; shuffling and sorting is done by the Hadoop framework before the reducers get their input. In reduce stage, once the reducer has got its respective portion of intermediate data from all the mappers, it performs user specified reduce function. The final reduce output is written on HDFS (e.g., HDFS://user/frank/wc_data1_output) as shown in figure 1.

B. YARN

YARN is essentially a distributed operating system that provides computational resources in the Hadoop cluster needed for running distributed applications. Apache Hadoop YARN Architecture consists of three main components: Resource Manager RM (one per cluster), Node Manager NM (one per node) and Application Master AM (one per application).

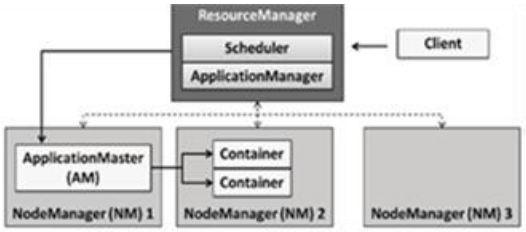

Figure 2. Overview of YARN Architecture

RM is the master of the YARN framework. It knows where the slaves (NM) are located and how many resources they have. RM has two main components: Scheduler and Applications Manager. Scheduler is responsible for allocating resources to the various running applications based on resource availability and the configured sharing policy. It also performs its scheduling function based on the resource requirements of the applications. Application Manager manages running the Application Masters in a cluster. It is responsible for accepting job submissions from the client, negotiating the first container from the Resource Manager to execute the application specific AM [14].

NM is the slave and offers some resources to the cluster. Its resource capacity is the amount of memory and the number of virtual cores (vcores). It takes care of individual nodes in a Hadoop cluster and manages application containers assigned to it by the RM. Container is a specified resource fraction (memory, CPU etc.) of the NM capacity on a host machine and many containers can reside on it [4]. How many number of cc can be run on a node, it depends on default resource calculator of capacity scheduler (default yarn scheduler). The DefaultResourceCalculator only checks memory amount specified by requests and the available memory of the node. By default, processing CPU resource is not considered.

AM is responsible for the execution of a single application. It asks for containers from the scheduler and executes specific programs on the obtained containers. It works along with the NM and monitors the execution status of tasks and progress for the application. Where, an application is a single job submitted to the framework. Each application has a unique Application Master associated with it which is a framework specific entity [14].

III. DETERMINING THE NUMBER OF AND WHY CC IS SELECTED FOR PERFORMANCE TUNING

TABLE I. MEMORY RELATED PARAMETERS THAT CONTROL THE NUMBER OF CC

Configur-ation File	Parameters' Name	Description	Default (MB)
yarn-site.xml	P1: yarn.nodemanager.resource.memory-mb	Memory that can be allocated for containers per node	8192
	P2: yarn.scheduler.minimum-allocation-mb	Memory allocation for every container request at RM	1024
	P3: yarn.scheduler.maximum-allocation-mb	Memory allocation per container for each resource request at RM	8192
	P4: Yarn.app.mapreduce.am.resource.mb	Memory per container for each MR AppMaster	1536
mapred-site.xml	P5: mapreduce.map.memory.mb	Memory per container for each map task.	1024

The table 1 describes some important memory-related parameters and their corresponding default values based on the machine specifications for determining the maximum number of cc per node in a cluster. According to the default settings in table 1, the maximum number of cc is 6 (available RAM for tasks =p1-p4=8192MB-2048MB=6144MB, cc= 6144MB/1024MB based on p5, where p4 is set based on increment of p2 if the p4 is greater than p2). Therefore, the number of concurrent container per node is within the range of 1 to 6. Where, we need to select how many number of cc per node should be run for improving the performance of Hadoop. Therefore, we experimented on cc per node. According to experimental result, tuning cc choice can improve performance of map stage significantly. In other word, tuning cc allow to take advantage of node performance leading to faster execution time of MapReduce applications. Therefore, other performance impact parameters relating to Hadoop 2.x (such as the parameters in [12] collected from many research papers) can be tuned to achieve better performance improvement based on proposed system.

IV. EXPERIMENTAL EVALUATION

This section presents the experimental environment and the experimental results obtained for various workloads.

Our motivation for this experiment is two-fold: to measure the execution time difference of various cc (on the same HDFS block size) per machine instance and to measure the performance impact of HDFS block size for possible cc per node, where possible cc is within the range of 1 to 6.

A. Experimental Setup

In our experiment, Hadoop-2.7.2 is used and is deployed on virtual machine (vm) using VMware Workstation 12 Pro. This machine is setup with 8G RAM and 4vcores and 200GB disk space. On this vm, pseudo-distributed mode with YARN is setup. This mode is a single-node cluster where all the Hadoop components will run on a single machine and is mainly used for testing purpose.

In this paper, we use a single machine instance and run only one MapReduce application (i.e., concurrently running application is one) at each test run. Thus, cc is calculated for running map tasks per application.

B. Experimental Benchmark

TABLE II. EXPERIMENTAL BENCHMARK USED IN THE EXPERIMENT

Benchmark MapReduce Program	Map Tasks or cc	Benchmark Dataset(MB)		
TeraSort	1, 2, 3, 4, 5, 6	***Data Generator***	***BS=128***	***BS=256***
		TeraGen	128, 256, 384, 512, 640, 768	256, 512, 768, 1024, 1280, 1536
Sort	1, 2, 3, 4, 5, 6	Random Writer	128, 256, 384, 512, 640, 768	256, 512, 768, 1024, 1280, 1536
Wordcount	1, 2, 3, 4, 5, 6	RandomText Writer	128, 256, 384, 512, 640, 768	256, 512, 768, 1024, 1280, 1536

Table II lists the selected representative MapReduce programs and benchmark datasets generated by data generators used in our evaluation. In benchmark programs, different MapReduce applications with a variety of typical Hadoop workload characteristics are considered: WordCount (CPU-bound), Sort and TeraSort (I/O bound) [3]. These programs are taken from the Hadoop distribution and serve as standard benchmarks and they are commonly

used to measure MapReduce performance of an Apache Hadoop cluster [7, 8, 2].

For each MapReduce program, we use 128 MB (default) and 256MB to evaluate the block size effect. Therefore, for each MapReduce program, the corresponding data generators are used to generate six input datasets (in Table 2) for each block size that produce required number of map task based on the number of cc. That is, cc is set to 1, we run 1 map task on 1 data block.

C. Analyzing the Effect of Map Stage Elapsed Time (MSET) Based on cc and BS for Experimented Applications

TABLE III. MAP TASK ELAPSED TIME BASED ON CC AND BS

MapReduce Program	Block Size (BS)	Map Stage Elapsed Time (sec) Based on cc or Concurrent Map Tasks					
		1	*2*	*3*	*4*	*5*	*6*
TeraSort	128 MB	7.5	20	35	39.5	64	129
	256 MB	15	40.5	74	98.5	186	210
Sort	128 MB	5	24	42.5	54	68.5	89
	256 MB	8	52.5	86	102	144	178
WordCount	128 MB	19.5	26.5	41	52	67	83
	256 MB	35.5	45.5	72	91	117	152.5

Table III shows the map task elapsed time of each MapReduce program (TeraSort, Sort and WordCount) based on possible concurrent containers for the experimental cluster. Each program is run on both HDFS block sizes: 128MB (default) and 256MB; and collect Map elapsed time on the corresponding input data sizes for three workloads. For each test run, we run three times in order to consider the execution time variances and for each test run, we choose the median values for better performance measurement

TABLE IV. PERFORMANCE IMPROVEMENT % BASED ON OPTIMAL CC FOR EACH PROGRAM AND BS

Map Reduce Program	Optimal cc	Improvement % for Input Data=15G	
		BS=128MB	*BS=256MB*
TeraSort	1	52.3322	59.15813
Sort	1	47.90419	64.23517
WordCount	4	4.076878	18.05475

Table IV shows the performance of map stage for all three MapReduce applications in terms of map stage elapsed time. In order to compare the performance difference between optimal cc based on BS and Hadoop's default cc. It is complex to explicitly/directly compare the differences because different cc run on different data size. (e.g. map task elapsed time on cc of 4 is longer than that of 2, however cc of 4 can run two times of input data size on cc of 2). To clearly compare the map elapsed time differences, the case of input data with 15GB is considered and the map stage elapsed time is computed based on the relation of map elapsed time on the number of cc along with block sizes in table 3 without running the workloads. The optimal number of concurrent containers per node is 1 for both TeraSort and Sort and 4 for WordCount.

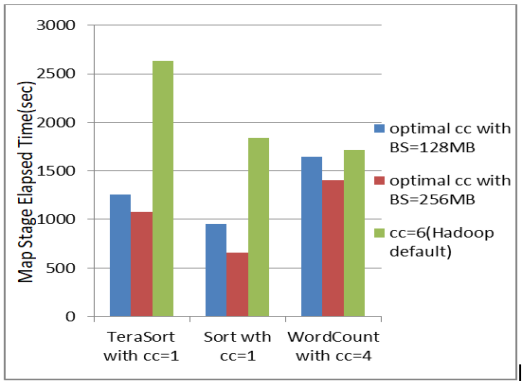

Figure 3. MSET Comparisons Between Hadoop Default cc and Observed Optimal Number of cc

This figure briefly describes the Map stage elapsed times for performance comparisons between observed optimal cc based on two block sizes(one is Hadoop default and other is suitable block size based on rule-of –thumb) over default cc. For all three workloads, optimal number of cc with block size of 256MB for each map task result in optimal performance with minimum the Map stage elapsed time, while default block size of 128MB is suboptimal.

D. Result and Discussion

Based on the computed results based on input data size of 15GB, the optimal number of concurrent containers per node is 1 for both TeraSort and Sort and 4 for WordCount. Where, sort and terasort require more disk other than CPU as these resources are shared among tasks. Consequently, the default based cc of 6 per disk on a single node cluster results in poor performance. Therefore, the optimal performance on Map stage can be obtained when the number of cc is one per disk. TeraSort program with optimal cc of 1 improve the performance of map stage by 52.3322% and 59.15813 with HDFS block size of 128MB and 256MB , respectively. Sort program with optimal cc of 1 improve the performance of map stage by 47.90419

% and 64.23517 % with HDFS block size of 128MB and 256MB, respectively.

However, in the case of WordCount, the optimal performance on map stage can be reached when the number of cc is 4 containers per node because the workcount is mostly CPU bound and the experimented machine instance has 4 vcores. For CPU bound job, using one map task per CPU can achieve optimal performance and for heavy disk or I/O, running only one map task per disk results in optimal performance in terms of map stage elapsed time. WordCount program with optimal cc of 4 improves the performance of map stage by 4.076878% and 18.05475% with HDFS block size of 128MB and 256MB, respectively. The optimal number of concurrent containers per node varies depending on the applications characteristics, such as heavy I/O and heavy CPU. Hence, prior to running MapReduce jobs, the cluster performance can be optimized via improving the node performance on map tasks by tuning the number of map tasks or cc per node along with suitable HDFS block sizes.

V. CONCLUSION

Performance optimization of MapReduce applications via Apache Hadoop Parameter configuration is still an open issue for researcher. This work investigates the performance limitation of Hadoop framework depending on container based static resource management and analyzes the performance improvement. The experimental results showed that different settings of concurrent containers running map tasks on the same default HDFS block size lead to different performance. In addition, we also observed that the performance can be further improved by tuning CC along with suitable HDFS block size. To show the performance differences among cc (along with two HDFS block sizes) per node, we measure the performance in terms of map stage elapsed time with example input data of 15GB for three MapReduce applications. This work achieves improvement of 59.15813, 64.23517 and 18.05475 in map stage elapsed time for the MapReduce applications: TeraSort, Sort and WordCount, respectively when compared to running them with Hadoop-default settings. In the future, the dynamic optimization on cc is performed by classifying the job types on fully-distributed mode by running multiple jobs concurrently.

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Mongodb on Cloud for Weather Data (Temperature and Humidity) in Sittway

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Abstract

The environmental conditions play the major effects on human beings and the weather parameters are very important roles in our daily life. Many research efforts have paid to solve the environmental problems. So the collecting of data about different parameters of the weather is necessary for planning in home and environment, and the database of weather parameters become more important for living things. In this work, two weather parameters: temperature and relative humidity in Sittway township, Rakhine state have been measured by Raspberry Pi 4 with DHT22 sensor for solving the environmental problems. The collected data from the system have been stored and transmitted to the cloud by MONGODB with Java Script programming language and then the comparison of the data collected from the sensor and the Mongo database on cloud has been made for the accuracy of the project.

Keyword — DHT 22, IoT, Momgodb,

I. INTRODUCTION

Weather condition plays an important role in our daily life as weather and climate are the most ubiquitous factors for home and environment planning. Moreover, the tremendous development of Internet nowadays made possible to monitor weather conditions and collect the respective data in-situ. All the objects, sensors and devices can be linked through Internet to share and analyze the data collected at various locations.

Most Internet applications focus on providing information for human beings. IoT is Internet of Things, known as M2M between smart devices that collect data, relay information to the others, process the information collaboratively, and take action automatically. IoT can be much more extensive in predicting and knowing the weather conditions in particular place by connecting the weather station to the Internet. IoT is a system consists of things attached with sensors, connected to the internet via wire or wireless network. In this paper, the attempt to support the solving the environmental problems and the real time weather conditions have been made to get the smart
system for human beings and, living things or non-living things[1][6].

DHT22 is a digital sensor with built-in analog-to-digital converter. It consists of both temperature sensor with negative temperature coefficient and humidity sensor. So, it has been used to detect the temperature and relative humidity of the desired environment. Raspberry Pi 4 reads the output from DHT22 first. And then the data collected are continuously stored in SD card of the Pi-board. By using MONGODB, these data are transmitted to the cloud to make the sharing data with the others, and the predicting the weather condition of specific area[2][8].

II. RELATED WORK

S. Adnan et.al proposed “Low Cost Embedded Weather Station with Intelligent System” using reflective optical sensor, 1-turn continuous potentiometer, low-power linear active thermistor. All sensors being used were basic type sensors, so the cost of the system was reduced. And “Design of Weather Monitoring System using Arduino Based Database Implementation” to measure and store temperature, humidity and wind speed was made by N. M. Sarmad and F. H. Forat. And then N. Ahmad et.al proposed “Design and Construction of Digital Anemometer”. A three-cup anemometer was designed to reduce the measurement errors in this work and microcontroller was used to make it digitalized.

III. BACKGROUND THEORY

Weather forecasting normally tells people the weather conditions for a certain place and a certain period of time. However, the forecasting sometimes

cannot predict precisely, especially in a particular case. For example, strong wind during winter would make the actual feel temperature much lower than what it is. To support the solving these problems, the weather forecasting station has been built and tested. Control system, embedded system and wireless communication are essential parts in weather station[9][11].

A. Control System

Measuring, comparing, computing and correcting are four functions in control system. The measuring is completed by detector, transducer and transmitter. Comparing and computing are within the controller and the correcting is with final control element.

B. Embedded System

Embedded system consists of hardware, software and other parts to perform specific function. Personal computer has general purpose and is able to do many different things. The embedded system is the system within a larger system. Modern cars and trucks contain many embedded systems. One embedded system controls anti-lock brakes, another monitors and controls vehicle's emission and a third displays information on the dashboard.

C. Wireless Communication

The transfer of information between two or more points is called wireless communication by electromagnetic waves. Wireless sensor networks are responsible for sensing noise, interference and activity in data collection networks. This allows us to detect relevant quantities in monitor and collect data, and to perform decision-making functions.

Wireless data communications are used to spin a distance beyond the capabilities of typical cabling in point-to-point or point-to-multipoint communication, to provide a backup communication link in case of normal network failure, to link portable or temporary workstations, to overcome situations where normal cabling is difficult or financially impractical, or to remotely connect mobile users or networks[5][10].

IV. EXPERIMENTAL DETAIL

The proposed system consists of (i) temperature sensor, (ii) humidity sensor, (iii) data input from hardware sensors and (iv) data output to storage SD card and to the cloud for sharing with the others.

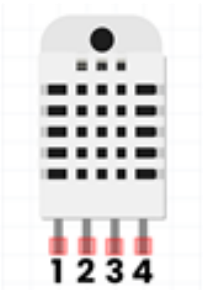

Figure 1. DHT22

Figure 2. Raspberry pi 4

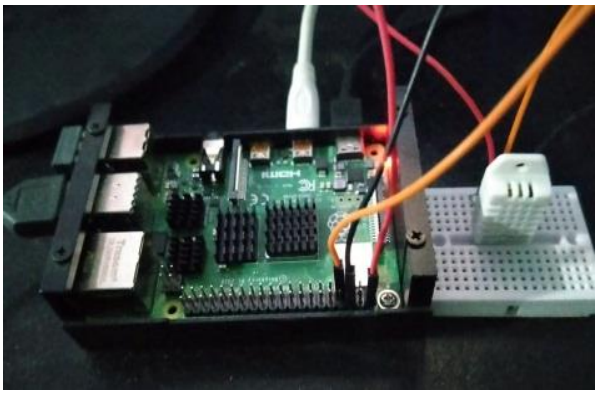

Figure 3. Proposed system

In this proposed weather data collected system, DHT22 has been used for temperature and humidity of the environment. It is a digital sensor with an inbuilt analog-to-digital converter (ADC) and the data can be transmitted through wire up to 20 m away from Raspberry pi. It consists of a humidity sensing component, a NTC (negative temperature coefficient) temperature sensor and an IC on the back side of the sensor. It is necessary to put on 10 kΩ resistor between pin-1 and pin-2 of DHT22.

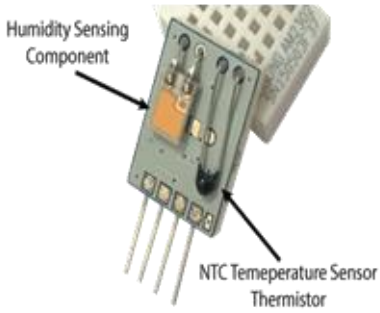

Figure 4. Internal configuration of DHT22

The weather data such as temperature and humidity in Sittway, have been collected by using DHT22 with the aid of Raspberry Pi 4 by Java Script. The first part creates the database ("test") and the collection ("dht") which apply the connection to the cloud. For the data sorting on CLOUD, the first creates the variable "insert" for temperature and humidity. When the term "data inserted" appears, the data value can be inserted, and it is necessary to create ID on cloud. The duration time between one-data and another is 10 s[3][7].

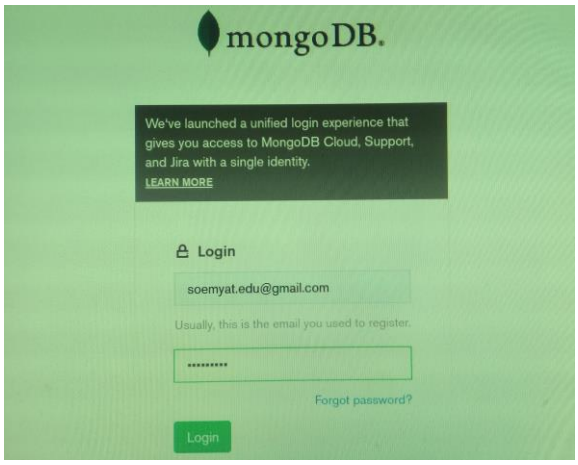

Figure 5. Email address of MongoDB to Cloud

MongoDB is a cross-platform, document oriented database that provides, high performance, high availability, and easy scalability. It works on concept of collection and document. MongoDB is also a NoSQL type database. NoSQL is not a relational database. It provides more flexibility, since all records are not restricted by the same column names and types defined across the entire table. Fig. 5 shows email address of MongoDB "soemyat.edu@gmail.com" and its password "005350soe".

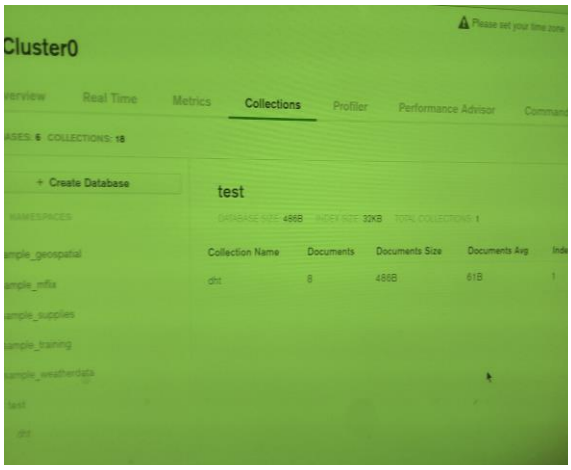

Figure 6. Create "test" database on cloud

Database is a physical container for collections. Each database gets its own set of files on the file system. A single MongoDB server typically has multiple database. To store the weather data, "test" database must be created on cloud. Collection is a group of MongoDB documents. It is the equivalent of an RDBMS table. A collection exists within a single database. In this work, "dht" collection has been made under "test" database.

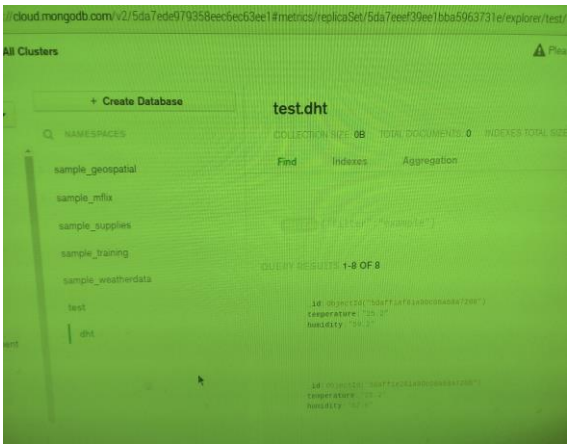

Figure 7. "dht" collection of "test" database

In the proposed work, "dht" collection of "test database" has been created for temperature and humidity data in Sittway. MongoDB can be used for "BIG DATA" in its collection of respective database. It is a document database in which one collection holds different documents. Number of fields, content and size of the document can differ from one document to another, and then data is stored in the form of JSON style documents.

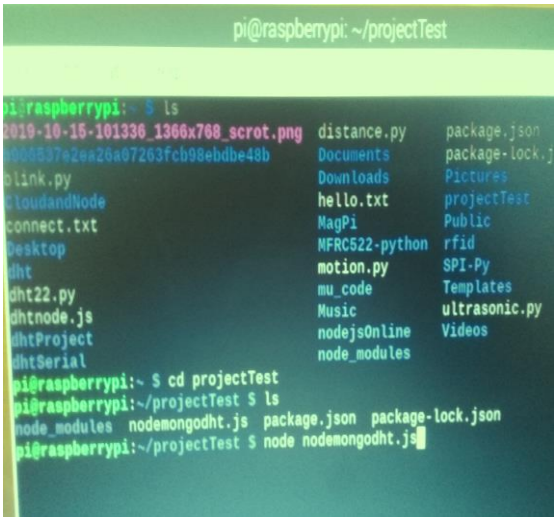

Figure 8. nodemongodht.js

For the running program of node.js, it is necessary to make the command "node nodemongodht.js". In this work, there are three programs such as "nodemongodht.js, "package.json" and "package-lock.json" in "projectTest" directory.

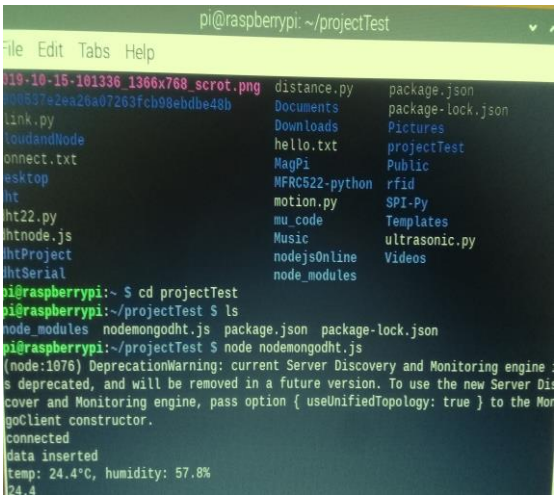

Figure 9. Output data of the system

As shown in Fig. 9, the first data output of temperature is 24.4ºC and its humidity is 57.8%. The second data will be produced within 10 s after the first output. The temperature and the humidity data of the desired environment for specific duration time are continuously collected by the proposed system.

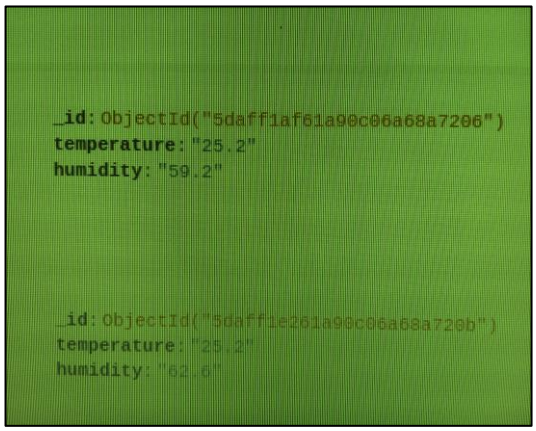

Figure 10. Temperatures & Humidities on

It is now seen that each document is printed in JSON style. The temperature and humidity data in Sittway, Rakhine state, transmitted into the cloud are shown in Fig. 10.

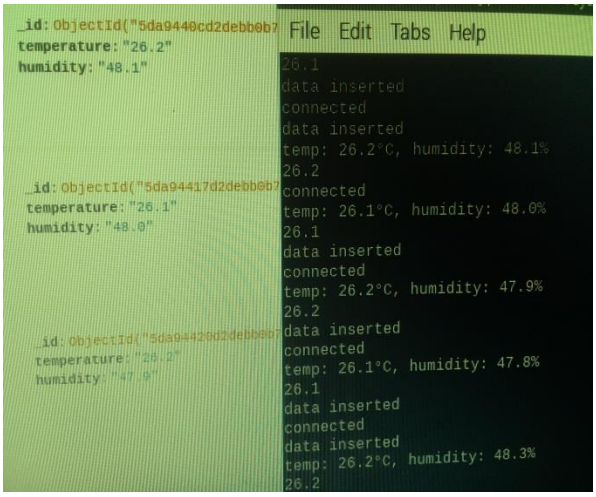

Figure 10. Comparison of sensor outputs and data on Cloud

Fig. 11 shows the comparison of DHT22 sensor outputs and data on cloud for the accuracy of the project.

Sensor Data

data inserted
connected
temp: 26.2 °C, humidity: 48.1% **(*)**
data inserted
connected
temp: 26.1 °C, humidity: 48.0%
data inserted
connected
temp: 26.2 °C, humidity: 47.9%

data inserted
connected
temp: 26.1 ^{o}C, humidity: 47.8%
data inserted
connected
temp: 26.2 ^{o}C, humidity: 48.3%
data inserted
connected
temp: 26.2 ^{o}C, humidity: 48.1%
data inserted
connected
temp: 26.1 ^{o}C, humidity: 48.3%
data inserted
connected
temp: 26.2 ^{o}C, humidity: 48.0%

Cloud Data

_id:ObjectId("5daff1af61a90c06a68a7206")
temperature: "26.1"
humidity: "48.0"
_id:ObjectId("5daff1af61a90c06a68a720b")
temperature: "26.1"
humidity: "48.3"
_id:ObjectId("5daff1af61a90c06a68a720c")
temperature: "26.2"
humidity: "48.0"
_id:ObjectId("5daff1af61a90c06a68a7205")
temperature: null
humidity: null (*)
_id:ObjectId("5daff1af61a90c06a68a720a")
temperature: "26.2"
humidity: "48.1"
_id:ObjectId("5daff1af61a90c06a68a7207")
temperature: "26.2"
humidity: "47.9"
_id:ObjectId("5daff1af61a90c06a68a7208")
temperature: "26.1"
humidity: "47.8"
_id:ObjectId("5daff1af61a90c06a68a7209)
temperature: "26.2"
humidity: "48.3"

As shown in the data list on Mongo-cloud, the identification numbers (Object Id) are not in ascending list. So, it is necessary to get the careful data sorting for the comparison between the sensor output data and the data on the cloud. In the above description, every row has a unique objectId. The output clearly shows that all of the documents are printed in JSON style. JSON is a format called JavaScript Object Notation and is just a way to store information in an organized, easy-to-read manner. The comparison of sensor-data and cloud-data is shown in Fig. 12 and Fig. 13. According to the data comparison, the first data bit cannot reach onto the cloud (*), and the data sorting on the cloud must be taken carefully at the beginning.

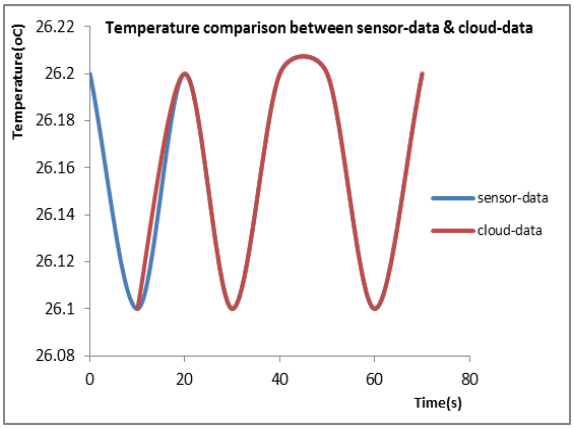

Figure 12. Temperature comparison between sensor-data and cloud-data

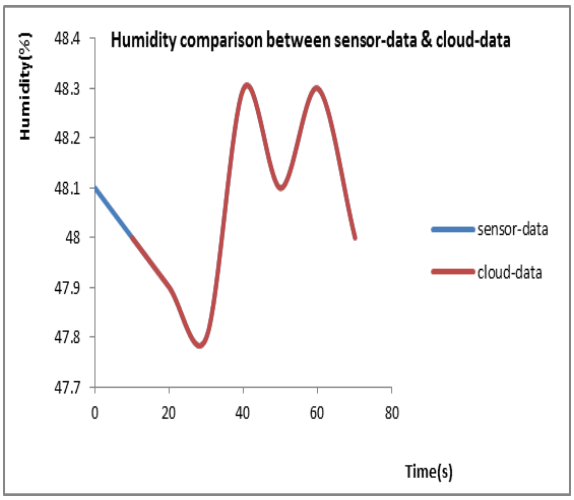

Figure 13. Humidity comparison between sensor-data and cloud-data

"test" database with "dht" collection on cloud for temperature and humidity of Sittway, Rakhine state is shown in Table (1) and Fig 12. When the temperature is less than 26^{o}C and the humidity is greater than 80%, there will be expected to rain in 30 minutes. So the weather condition of Sittway in this period is fine[4][12].

TABLE I. TEMPERATURE AND HUMIDITY IN SITTWAY

Temperature & Humidity in Sittway			
Date	Time	Temperature(° C)	Humidity(%)
25-10-2019	13:36:00	26.2	48.1
25-10-2019	13:36:10	26.1	48
25-10-2019	13:36:20	26.2	47.9
25-10-2019	13:36:30	26.1	47.8
25-10-2019	13:36:40	26.2	48.3
25-10-2019	13:36:50	26.2	48.1
25-10-2019	13:37:00	26.1	48.3
25-10-2019	13:37:10	26.2	48

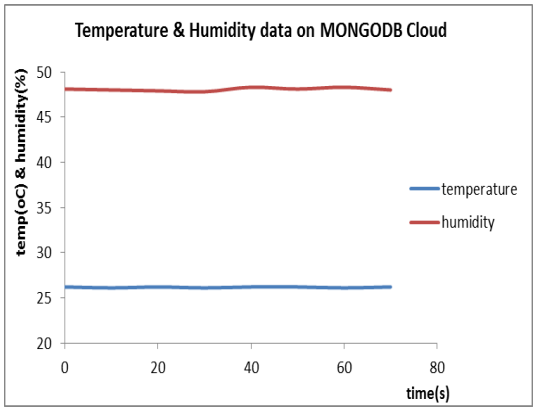

Figure 14. Temperature and humidity on cloud

V. CONTRIBUTION

The proposed system has been designed to create the low cost weather station to get the information of real time weather condition and easy to install to achieve the weather data of a specific area. And then the data can be used to share to the others by using MONGODB with node.js.

VI. CONCLUSION

Weather prediction is a very important factor, which forecasts the climate in a region based upon the values of weather parameters. So the calculated results from this system can be used in forecasting the weather of that locality for a period of time. This research makes the understanding concepts of humidity sensor and temperature sensor of DHT22 according to the construction of them, and then how to create the MONGO database on cloud of the weather parameters of the specific area. The data from sensor are transmitted to sever where they can be viewed globally which will be easily accessible to everyone. All the weather parameters were successfully displayed via MONGO database which are accessible by both administrator and users. Because there is no concept of relationship in MongoDB, a document database, in which one collection holds different documents and it can deal with big data. The comparison between sensor data and cloud data has been made for the determination of the accuracy of mongo-node.js on private-cloud. The system can make to solve the environmental problems due to the weather condition for living-things and non-living things.

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Preserving the Privacy for University Data Using Blockchain and Attribute-based Encryption

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Abstract

An effective IT solution is required for the administrative processes which is the core of university. The tamper resistance, transparency and auditability for university management data are very important to avoid the corruption. Blockchain occupies the immutability and irreversibility properties thus, it becomes a potential solution. Regrettably, there are some challenges such as privacy concern and limited storage exists in blockchain technology. In this work, a blockchain-based university data storage model is proposed and such barriers of blockchain are handled. The proposed model achieves the tamper resistance, transparency and auditability feature by using blockchain technology. CP-ABE and other cryptographic techniques are used to provide the fine-grained access and privacy preserving facilities. The security analysis is performed and shows that our approach is tamper resistant and provably secure for privacy.

Keywords—*data security, tamper resistance, blockchain, privacy, fine-grained access*

I. INTRODUCTION

In the premise of industrial 4.0 revolutions [1], human society will be stimulated with the higher education which is an essential factor for human development. Thus, a large number of information systems such as office automation system, teaching system, administrative system, personnel system and asset system are built in universities to improve the efficiency of staffs [1]. This work will make dialectical, complicated an interesting prospects of human centric characteristic. The research, services and teaching will be changed to different ways and new forms of universities will be emerged in the fourth industrial revolution. Thus, the tamper resistance, transparency and auditability for university management data are very important to avoid the corruption. Then again, the industrial 4.0 also takes the popular technology, blockchain which occupies the transparency, immutability and cryptographic verifiability properties. Thus, blockchain become a candidate solution for university data.

Contrary to the expectation, the blockchain technology also contains some obstacles in using for university data. One of the obvious requirements for university data is fine grained access that allows the only authorized user to access the certain data. For instance, if the university data of all departments are transparently stored on blockchain, any faculty or everybody who can access the blockchain can access any document. However, the blockchain stores cryptographically verified data; it does not encrypt the data at all [2]. In some private blockchain system like Hyperledger Fabric [3], the participation in network can be regulated, however, the university data still needs to allow only certain users to access a specific data. Consequently, the privacy issue has to be handled. The append-only property of blockchain becomes a barrier for a user revocation feature which allows eliminating permission of the access on the university data for specified individuals. The explosive growth of the university data causes the available issue for the limited blockchain storage. Thus, how to use blockchain technology for university data as an underlying mechanism is still an issue.

To provide the confidentiality and to preserve the privacy, encryption is a promising way. However, there exist some traditional technology such as password based and classical public key encryption based approaches, the individual user needs to maintain too much secret information (decryption keys or passwords) for accessing the multiple files. The attribute-based encryption (ABE) [4] uses the user attribute set as the public key instead of using random string as the public key. The further development of ABE encrypts the message with an access policy and it is called Ciphertext Policy Attribute-based Encryption (CP-ABE) [5]. In CP-ABE scheme, only the user who

has the attributes that meet the access policy requirement can decrypt the encrypted message thus, CP-ABE can support more efficient access control mechanism.

In this work, the blockchain technology, cloud storage and CP-ABE encryption technique are used to preserve the privacy for the university data. The blockchain technology is used to support the tamper resistance, auditability and transparency of university data. The cloud storage is used to overcome the availability issue of blockchain and CP-ABE is used to support the fine grained access on university data. Thus, our approach requires only one secret key per user while other systems such as password based systems require multiple passwords for multiple files. Similarly, our approach requires encrypting the data only one time while public key crypto systems require multiple times for encryption for a document for multiple users. At the same time, our approach can support tamper resistant, auditability and transparent property for university management data.

The followings are the rest of the paper. Section 2 discusses about some existing related works. Section 3 discusses some technology and knowledge that support in developing our system such as blockchain technology, CP-ABE encryption and other cryptographic primitives. The detailed of our proposed approach is discussed in section 4 and security analysis is performed in section 5. Finally, the paper is concluded in section 6.

II. RELATED WORKS

Blockchain is used as the storage for diverse items such as academic work, attendance, certificates and awarding of a university degree in the education purpose [6], [7]. The distributed and transparent properties of blockchain are used to reduce the fraud in academic items such as certificates and informal context such as misrepresentation of knowledge, background and skill [8]. Most of these existing works are trying to use some properties of blockchain such as transparency and there are some challenges such as confidentiality, privacy and availability for using blockchain.

In [9] the fine grained access control for encrypted data is firstly developed with a variant of ABE. However, this approach cannot support complex policies require for university data. To support such sophisticated policies, CP-ABE which describe the users with various attributes is more suitable. The CP-ABE algorithm is firstly introduced by Bethencourt et al. in [6]. They used access tree in encrypting the message and "Lagrange Interpolation" is used in decryption. Then, a comparative attribute-based encryption is proposed in [10] and the proposed method is illustrated with an example of telemedicine. Moreover, an implementation of Functional Encryption is proposed in [11]. Most of the existing works tried to use attribute-based encryption for healthcare data.

There is no universally accepted standard definition for the term "privacy". Privacy encompasses with several concepts such as anonymity, pseudonymi ty, unlinkability, unobservability and revocability of consent. The blockchain support most of these concepts, thus, we try to use the benefits of blockchain technology for university data and the cloud technology and CP-ABE technology is used to handle some blockchain issue for using in university data storage.

III. PRELIMINARIES

A. Blockchain

The blockchain technology is introduced by Satoshi Nakamoto with the crypto-currency called Bitcoin [2]. Actually, the blockchain is a distributed database and each data storage structure called block links each other as a chain [12]. A block is a special storage structure of blockchain and it maintains the hash value of the previous block to form a chain. By forming a chain, the block is immutable to modification [13]. The block also maintains other items such as payload, timestamp and signature of the contributor. Payload of the block may vary according to the various applications. The payload can be any asset or item such as the content of the transaction, an address pointer of the original data or some other information. The timestamp is used to order the blocks in chronological fashion. The signature of contributor shows the generator of the block. Generally, blockchain network includes two main entities, miners who produce new blocks and verifiers who verify the new blocks. In generating the new block, consensus mechanism is usually used. In this work the metadata will be stored on blockchain as verified data.

There are three categories of blockchain in general. They are permission-less blockchain also called public blockchain, permission blockchain also called private blockchain and consortium blockchain. The permission-less blockchain is public in nature and everyone can participate in the blockchain system. Bitcoin is a good example for permission-less blockchain system. In the group of permission blockchains, the access right and participation on

blockchain network is controlled by an organization in private blockchain while several organizations manage the consortium blockchain. The permission blockchain (private blockchain or consortium blockchain) is suitable to store and manage the university data according to the architecture of our work.

B. Ciphertext-policy Attribute-based Encryption

The ciphertext-policy attribute-based encryption (CP-ABE) [5] is a one-to-many encryption scheme. CP-ABE allows the multiple users to access the encrypted data. CP-ABE use a set of attribute in identifying the user with the decrypting key called CP-ABE private key. By using a list of attribute which correspond to the authorized users, the data owner can specify various access policy. The policy is then embedded into the encrypted data. For example, the policy can be expressed as follow.

Policy P = "(Rector) OR (Head_of_Department) AND (Department_1)"

If the user processes the attributes 'Rector' or 'Head of Department from Department 1', the ciphertext can be decrypted. Otherwise, the data cannot be decrypted. Generally, there are four main steps in CP-ABE scheme. In the first step, a master secret key MSK and a public parameter PK are generated and this step is called setup phase. The public parameter PK contains the generator g, g^{β}, and an efficiently computable symmetric bilinear mape $(g, g)^{\alpha}$. The master secret key MSK contains the value β and g^{α}. The PK can be reveal publicly, and the MSK must be kept secret.

In second phase, the encrypting process is performed. The cipher text CT is output from the set of input which includes plaintext message M, public parameters PK and an access policy T in encryption phase. A set of Boolean formulas is used to create an access policy tree. The figure 1 illustrates the creating the access policy tree from the policy P.

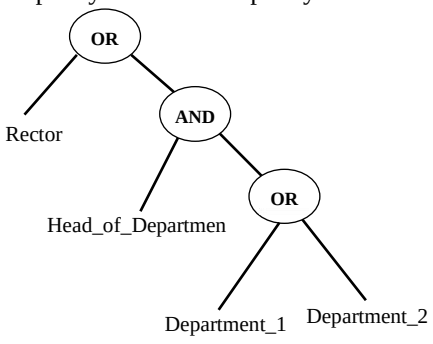

Figure 1. Sample access policy tree

The third step is the key-generation phase. The private key SK which associate to the set of user attributes set S is generated in this phase. The master secret key MSK, the public parameters PK and a set of user attributes S are taken by this process as input. That is why, the attributes are mathematically incorporated into the key.

The fourth step is the decryption phase. The ciphertext CT will be decrypted in the decryption phase, if and only if the access policy tree T is satisfied by theset of attributes associated with the private key SK.

C. Access Structure

Let a set of $n \in N^{+}$ be a set of participants $\{P_1, P_2, \dots, P_n\}$ and a collection $A \subseteq 2^{\{P_1, P_2, \dots, P_n\}}$ is monotone for $\forall B$ and C, and if $B \in A$, $B \subseteq C$, then $C \in A$. Then, an access structure is a collection A of non-empty subsets of $\{P_1, P_2, \dots, P_n\}$, i.e, $A \in 2^{\{P_1, P_2, \cdots, P_n\}} \setminus \{\emptyset\}$ [5].

The sets which are included in A are called as the authorized sets. However, the sets which are not included in A are called unauthorized sets. Attributes take the role of parties in our context. Thus, the access structure A will contain the authorized sets of attributes.

D. Bilinear Map

Let G and $G_T$as two cyclic groups of order p for some large primep. G is a group of points on an elliptic curve over F_P (namely the finite field mod p) is G and a subgroup of a finite field $F_{p^2}^{*}$ (namely the finite field mod p^2) is G_T.

Then, a map $e : G \times G \rightarrow G_T$ is called to be a bilinear map if it satisfies the following properties.

1) Bilinear ($(aP, bQ) = e(P, Q)^{ab}$ for all $P, Q \in G$ and $a, b \in Z_p$).
2) Non-Degenerate (the map does not send all pairs in $G \times G$ to the identity in G_T and observe that since G and G_T are groups of prime order, this implies that if P is a generator of G, then $e(P, P)$ is a generator of G_T)
3) Computable (there is an efficient algorithm to compute $e(P, Q)$ for any $P, Q \in G$)

IV. METHODOLOGY

This section provides the problem scenario of our work and the detailed description about our proposed model.

A. *Problem Scenario*

Generally, in a university, the role hierarchy of the employees may be in the form which is illustrated in Figure 2. If the data are stored in the same location or storage, everyone in the university (all faculties or departments or all participants) can access these data. In reality, there may be various attributes of employees in university environment; however, the Table 1 illustrates some sample attribute list for employees in university.

To support data privacy (authorized access), data protection mechanism which allows only the authorized employees to access the data is required. The authorized employees must occupy the attributes values which can satisfy the access conditions. Thus, the attribute values which are shown in Table 1 are used to generate the decryption keys (private keys) for each group of users.

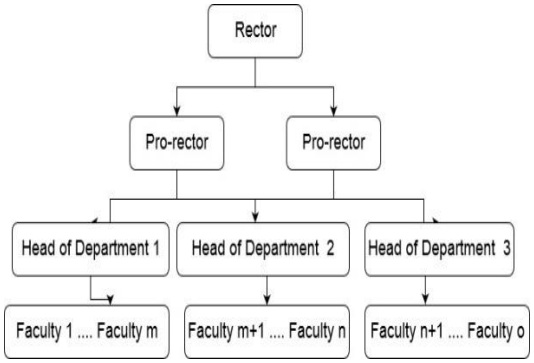

Figure 2. Sample hierarchy of employees in university

For instance, an examination data, S2_D3_ B5_Sem2_Lab_Marks_2016.doc is stored in the system. In this data the subject is S2, the department is D3, and the batch is B5, and the semester is the second semester Sem2. The employees associated with this batch should access it is confidential by this marks data.

Everyone can access this data if the data is stored in common storage. Password-based usage has operational difficulties such as maintaining many passwords although some password protection can be employed for protection. Thus, a new approach is required to store the data in the manner that the data is encrypted; the data is tamper resistant and the data must be accessible by only authorized users.

TABLE I. SAMPLE ATTRIBUTE LIST OF UNIVERSITY EMPLOYEES

Faculty	F1	F2	F3	F4	F5	F6	F7	F8
Department	D1	D1	D3	D1	D2	D1	D1	D1
Designation	Associate Professor	Assistant Professor	Assistant Professor	Professor	Professor	Associate Professor	Assistant Professor	Professor
Role	Lecture	Tutorial	Lab	Coordinator Theory	Lecture	In-charge D1 2018	In-charge D1 2Sem 2019	In-charge D1 2Sem 2018
ID	'ID = 1764'	'ID = 1760'	'ID = 1543'	'ID = 1729'	'ID = 1271'	'ID = 1270'	'ID = 1290'	'ID = 1250'
Batch	B1	B6	B5	B3	B2	B7	B4	B5
Subject	S1	S1	S2	S1	S3	S4	S5	S3
Semester	Sem2	Sem2	Sem2	Sem2	Sem1	Sem2	Sem1	Sem2
Year	2016	2017	2016	2017	2018	2018	2019	2018

B. Proposed method

Attribute-based encryption technique and blockchain propose a privacy preserving so as to support the data owner with the fined grained access control and tamper resistant storage. CP-ABE algorithm encrypts the actual data to support confidentiality. The encrypted data is stored on the cloud storage which can guarantee the availability. The metadata which represents the university data is permanently stored on the blockchain to provide a search and to obtain the tamper resistance property. Figure 3 represents the overall architecture.

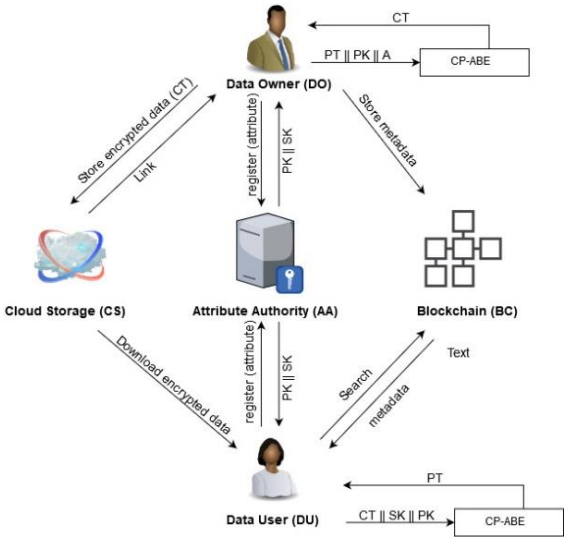

Figure 3. Architecture of the proposed model

There are five main entities that include in proposed model. The first entity is the data owner (DO). DO will access or store the university data. DO has full right to control the over the data, thus, DO will define the access policy to allow or disallow some accesses on data. The second entity is the data user (DU) and DU can access the data with the permission of corresponding DO. Typically, DU can be an entity from several sections. DU can search the information about data on blockchain via the metadata. Then, DU can download the encrypted data from cloud storage (CS). The third entity is the attribute authority (AA) and it is responsible for verifying the attribute of each user and issuing the corresponding key. The other entities are cloud storage (CS) which stores the encrypted data and blockchain (BC) which stores the metadata.

1) Setting the system

All participants in the system must register with the attribute authority to setup the system and perform the initial agreement to support the operations in the model. To start the agreement, AA performs the setup algorithm. The setup algorithm uses the implicit security parameter as an input and it produces a master key MK and the public parameter PK. Then, AA performs the key generation algorithm for each user with their attributes set S and master key MK. At the end of setting the system, AA distributes an appropriate private key SK to corresponding user.

2) Storing Data

The data owner (DO) starts the store operation by performing the encrypt algorithm with an access structure A over the universe of attributes and the sensitive information or message M. Therefore, users who occupy the required attributes set which can satisfy the access structure can decrypt the resulted ciphertext CT. On the other hand, the the access policy which is embedded in the ciphertext CT or CT implicitly maintains the authorized attributes A. Then, the hash code is generated from CT for integrity checking. CT is stored on cloud storage and the link to CT is received. The DO extracts the metadata and includes the hash code and link for search purpose within the system. Finally, DO stores the metadata and other required information about the encrypted data on the blockchain.

3) Retrieving Data

The DU starts the retrieve operation by searching the required information of encrypted data on the blockchain. DU uses the link from metadata to get the encrypted data from CS. DU checks the integrity of encrypted data with hash code obtained from metadata set. DU runs the decryption algorithm with his/her private key SK, the public parameters PK and the ciphertext CT.

If the attribute set S of DU satisfies the access structure A which is embedded in the ciphertext CT, then the algorithm can decrypt the CT and DU get the original message M.

V. SECURITY ANALYSIS

In this section, the security of proposed model is proven through the following cases of adversaries' attempts.

Case 1: Supposing that the data stored by some users in the cloud can be access by the adversaries, however the adversaries cannot alter or modify the data as well as cannot view the stored data. Even if the data

is altered, all the users can know such action. Thus, the data is tamper resistant and can be safely preserved.

Proof: The blockchain actually maintain the hash code of all uploaded data and the characteristics of blockchain ensure that the stored data cannot be modified. The data stored on blockchain cannot be altered or delete once confirmed. If the metadata in the blockchain is wanted to modify by the adversaries, they must try the extensive work to construct a new main chain. Such kind of action is neraly impossible. Moreover, if the data is tampered, it can be verified by comparison with the hash code preserved in the blockchain.

Case 2: Supposing that the adversaries attempt to cover the real content by preserving some piece of metadata which is different from the existing metadata, however, the early metadata which is stored on blockchain always exist and it is more legally effective, thus, the adversaries cannot defraud with the false metadata.

Proof: Generally, a judicial system will be needed to adjudicate with some evidence when two different metadata exist. However, no judicial evidence will be needed in our approach because each metadata has a timestamp that shows the time of preservation in the blockchain.

Case 3: Suppose adversaries can pinpoint all blockchain transactions and read the data. If he/she cannot understand the encrypted data, the adversaries will not be able to steal the sensitive information.

Proof: As all data are encrypted and then stored, and thus no one can see the real contents of the preservation as long as the private key is not compromised.

VI. CONCLUSION

In this paper, the CP-ABE is applied to efficiently control access right to achieve the benefits of blockchain technology for university data. All metadata that represent the university data are saved on an immutable and distributed data storage called blockchain. A user who doesn't have access right will not be able to view the data, and the stored data can verify that it had been modified. Thus, it provides a data integrity, authentication and reliable system. A CP-ABE based fine grained access control of data has also been presented in a university scenario. The university data can be stored at cloud storage and still accessible to only those users whose attribute values satisfy the access policy. The proposed model is part of our ongoing research thus, an experimental study will be conducted on the proposed model and evaluate the empirical result to improve the model in the future.

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Scheduling Methods in HPC System: Review

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Abstract

Parallel and distributed computing is a research area that is complex and rapidly evolving. Researchers are trying to get more and more methods and technologies in parallel and distributed computing. Most users who use Pay as you do go design, elasticity, and virtualization mobility and automation make cloud computing an attractive option to meet the development needs of some HPC users. Over the years, the research algorithm has also changed accordingly. The basic principles of parallel computing, however, remain the same, such as inter - process and inter - processor communication schemes, parallelism methods and performance model. In this paper, HPC system, scheduling methods and challenges are discussed and given some potential solutions.

Keywords**:** *parallel and distributed computing, high performance computing, methods, challenges*

I. INTRODUCTION

Scheduling technology is very important role in Information Era. Data input is processing in various ways and displaying. High Performance Computing (HPC) performs and demands many computers to work multiple tasks concurrently and efficiently. For a long time, the preparation of work for parallel computers has been subject to study. For a long time, the preparation of jobs since parallel computers has been subject to study. Job execution time is defined as the amount of processors allocated to it or requires users to provide estimated time for execution of jobs. The simulation software produces the estimation runtime attribute based on the specified range of estimation errors. The proposed method is to estimate and reduce execution time in a system.

Today, a super computer is and will always be a one-of - a-kind framework. Only a small number of users can use it. The operating mode can be compared to an experimental facility's exclusive use. A supercomputer does not usually have free resources. Typically the client has to wait, not the other way, to use a supercomputer program. During the past few years, supercomputers have been widely used in scientific research and industrial development. The architectures of such systems have varied during time. Today, to construct a real scientific grid, the main building blocks are mostly in place. The requisite communication quality is given by high speed wide area networks.

HPC system such as Grid computing [11], grid computing is a distributed network of large numbers of connected computers to solve a difficult problem. Servers and personal computers perform different functions in the grid computing model and are loosely connected to the internet and low-speed networks. Computers may connect through scheduling systems or directly. Grid computing involves dynamic digital organization (doctors, scientists and physicists), sharing of resources (reliable and unreliable collection of resources) and peer-to-peer computing. The grid also aims to provide access to computing power, scientific data resources and analytical facilities and is a grid computing challenge.

The key issue for high-performance computing is executing computational processes on a specific set of processors. Although the literature has also proposed a large number of schedules for heuristics, most of them target only homogeneous resources [19]. Future computing systems are most likely as the computational grid to be widespread and highly heterogeneous.

To achieve their performance goals, the development of research applications running on these systems has typically complicated new structural changes and new technology capabilities. Growing demand for new resources generation of HPC systems (computing nodes, memory, power, etc.) need to be supported by the ability to run more and larger applications with increased resource

utilization as well as HPC system job turnaround time.

The traditional purpose of scheduling algorithms is as follows: find a mapping of processor tasks and order tasks to fulfill a task graph and a set of computer resources: (i) task precedence constraints are met; (ii) resource constraints are met; (iii) a minimum schedule is established. A divisible job is a job that randomly divides the cab into any number of processors in a linear fashion. It applies to a parallel job perfectly: every sub-task can be performed in parallel on any number of processors.

II. BACKGROUND

Parallel systems are valuable resources, such as supercomputers, which are widely shared by user groups.

A. Parallel Computing

The early standardization on a single machine device, the von Neumann computer, gave a lot to the rapid penetration of computer systems into commerce, research, and education. A von Neumann pc incorporates a principal processing unit (CPU) connected to a storage unit (memory). The CPU executes a saved application that specifies a chain of study and writes operations at the memory. This easy model has proved remarkably robust. Its endurance over more than forty years has allowed the look at of such important subjects as algorithms and programming languages to proceed to a big volume independently of developments in laptop architecture.

High performance computing investigates the parallel algorithm and strengthens the parallel computing architecture. Parallel computation could be a kind of calculation in which at one time different enlightenment is performed. The goal is to extend computation speed to possibly understand complex assignments of computing. There are two different loads and loads of parallelism refers to pipeline, whereas space parallelism involves numerous synchronous computing processors.

B. Scheduler

Scheduling is defined as a method through which a task, specified by some means, is assigned to resources required to carry out the specified task. The research can be virtual computing components such as threads, processes or data streams, which in turn are built on hardware resources such as processors, network connections or expansion cards. The scheduling in CPU is difficult to control others. Multiple processors have to be scheduled in parallel computing and to manage the resources for all processors any overlapping of the resources to produce any conflicting results are needed. So the scheduling in multiprocessors is more complex than scheduling in a single processor unit. In HPC system, parallel computing is using to get high performance. In scheduling of multiple processors it should be ensured that any processor should not be overloaded and any processor should not be under loaded. There will be multiple processors; there will be multiple queues, so there is need of scheduling multiple queues simultaneously.

TABLE I. TYPES OF HPC SCHEDULER

Scheduler Name	Description
Slurm[9]	One of the HPC scheduler, a highly scalable, fault-tolerant resource cluster manager and massive computer systems job scheduler. A commonly used plugin-based job scheduler that provides several optimization options, either by adjusting the configuration parameters or by implementing new plugins and new scheduling and policy collection.
LSF[3]	It is a tool for task management, a job scheduler for high-performance distributed computing. It also allows users to access vast amounts of computing resources distributed around the network in large, heterogeneous distributed systems and can provide major performance upgrades to applications.
Loadlever	It schedules jobs and offers functions for the faster and more efficient production, submission and processing the jobs in a dynamic environment [21].
Moab [22]	A method of task planning and management for use on clusters, supercomputers, and grids. It can support a wide range of policy planning and fairness dynamic priorities, and substantial reservations.
Torque	It is an open source resource manager [23] that enables batch job control and distributed compute nodes.

The primary purpose of the job scheduler is to assign resources to work for users and ensure that jobs are operating at their highest performance. It prevents the overloading of a given compute node and puts jobs on hold before resources are available.

The scheduler's secondary purpose is to monitor usage to ensure fair distribution of HPC resources over time. To run multiple applications concurrently, HPC schedulers order to execution of batch jobs to achieve high utilization while controlling their turnaround times. In HPC schedulers, the balance between utilization and turnaround time is controlled by the scheduler prioritization system and the scheduling algorithms.

Turnaround Time: *Time from submission to completion of process*

Production HPC system uses different workload manager that combine scheduling and resource management. The following table 1 is most popular schedulers.

Modern HPC tools usually consist of a cluster of computing nodes that provide the user with the ability to coordinate tasks and substantially reduce the time it takes for complex operations to be performed. Usually, a node is defined as a discrete unit of a computer system running its own operating system case. Modern nodes have several processors, also referred to as Central Processing Units or CPUs, each of which contains multiple cores that can process a separate instruction stream.

All things considered, in spite of which strategy is used, the idea of a high-performance system is stable. The management of a high-performance system (referred to as part of a single gadget or multi-computer cluster by multiple processors) is treated as a single computing commodity, placing demands on unmistakable hubs. The HPC framework could be a partitioned unit created and actualized unequivocally as an effective computing device.

C. Time-sharing and Space-sharing

Time sharing refers to any scheduling approach that allows others to preempt and restart threads later during execution. The amount of jobs that each processor can perform at the equal time is defining as the level of multi-programming. Future HPC systems will be much noisier, with much greater competition and heterogeneity, requiring the use of new, asynchronous programming models [5].

Space-sharing procedures only offer a string that uses a processor until its execution is complete or the most extreme period of time has been reached and the string is done. Space sharing techniques [14] control time by putting that work in a line and at the same time running all its strings when discharged from that line.

D. HPC Scheduling

Parallel computing has become most important issue right this time but because of the high cost of computer it is not accessible for everyone. Cluster is only technique that provides parallel computing, scalability and high availability at low cost, in fig 1. Cluster collection provides high-performance computing, while individual computers work to solve a problem at the same time. Clusters are designed as they provide computation and availability of high performance over a single computer. A cluster is a group of connected devices, such as computers and switches that function as a single system together. Each and every node of a cluster is associated with a single system. Each and every cluster node is either connected by wire (Ethernet) or wireless that transfers data between nodes. A strong cluster provides distributed computing consisting of standard desktop computers linked through a network like Ethernet. Linux operating system can be used to control the cluster. In this way, we can build high performance computing (HPC) at low-cost price.

Scheduling method is mainly divided into two types: static and dynamic. With static scheduling, the consideration concerning processor placement and task assignment is made at the onset of the job execution. Static policies have low run-time overhead and the scheduling costs are paid only once, during the compilation time, for instance. As a consequence, static scheduling policies can afford to be based on sophisticated heuristics that lead to efficient allocations. Parallelism can be well exploited by spreading different tasks over different processors statically. Static scheduling needs to know in advance detailed information about the job's run-time profile which is relied on the input data, static scheduling carries some degree of unsure. The consequence may be an unbalanced load resulting in longer parallel execution time and low utilization of the network.

Dynamic scheduling, during execution, the number of processors assigned to a job can vary. The allocated processors are also assigned tasks during the execution of a job. In both their advantages and disadvantages, dynamic scheduling policies complement static policies. Typically, complex policies generate high overhead run-time, which can lead to performance degradation. But there are dynamic scheduling processes and adaptive behavior, which results in a high degree of load balancing.

HPC scheduling research [7] relates to how such research should be performed, because such research requires information, methodology, and resources that are not always available. Some of three basic methods for job scheduling research:

- Theoretical analysis
- Real system experiments
- Simulation

Theoretical Analysis

By defining boundary cases reflecting the best and worst-case results, algorithm behavior is analyzed. This approach may provide insight into the actions of the algorithm, but may not reflect the performance anticipated when performing a real workload. In HPC method, percentage of execution time spent in inter-process communication, congestion of memory bandwidth and performance of FLOPs are evaluating various ways to determine the degree of performance degradation when running practical HPC workloads.

Real System Experiments

Simple measurements of the behavior of the algorithm with a real system and workload still involve several different experiments [3] to be performed, which can be difficult for several reasons. However, a single experiment only produces evidence for one workload and process state, which is usually not enough to rationalize the general case or test a general hypothesis. Eventually, the conditions of workload are difficult to control, so evaluating specific scenarios can be challenging.

Simulation

A process is emulated, the emulated system runs a batch scheduler, and the scheduler receives a synthetic workload. This method allows the development of various experimental scenarios and the execution of large-scale experiments to produce information that enable general conclusions to be induced. Nevertheless, their findings are only true if they are indicative of the recreated workloads, processes and scheduling behaviors.

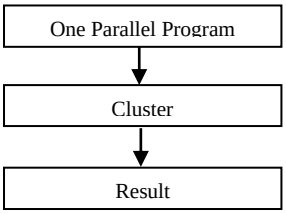

Figure 1. A cluster used to run a parallel program

No interaction with the administrator is needed through the usage of HPC program, jobs are completed sequentially without delays created by human interaction, which saves processing time this is generally wasted with human interaction. An application usually uses a parallel algorithm to run on a high performance cluster. A big task can be divided into several sub-duties, running on cluster-based exclusive nodes. The knowledge collected, resulting from the sub-duties, is translated into the special challenge's quit end result.

E. Need of Job Scheduling

Through a planning system responsible for defining available resources, a task is allocated to tools: matching job requirements with resources, making work ordering decisions and goals. Usually, the use of HPC resources is strong. In this manner, assets are not quickly accessible and employments are lined for future execution time until execution is regularly very long (numerous generation frameworks have and normal delay until execution of >1h), employments may run for a long time (a few hours, days or weeks).

Job Steps

A user job enters a job queue, the scheduler (its technique) chooses on begin time and asset properties of the work. The occupations can come at the shape of batch or interactive type.

- A batch job is a script is used to submit the job. A shell script may be a given list of shell commands to be executed in a given arrange.
- Today's shells are so modern simply can make scripts that are genuine programs with a few factors, control structures, circles.
- An interactive job is typically an assignment of one or more nodes where one of the cluster nodes receives a shell from the user.

Job scheduling is a mapping mechanism from the tasks of users to the proper selection and execution of resources [1].A scheduler may aim at one or more goals, for example: minimizing throughput, minimizing wait time, minimizing latency or response time or maximizing fairness [2].The following job scheduling algorithms are: First come first serve (FCFS), Shortest first job (SJF), Round - robin (RR), Priority scheduling, Min - min, Max - min, Genetic algorithm (GA) and so on which are using for job management.

A good scheduling algorithm allocates suitable resources to workflow tasks in order to complete the application and at the same time satisfy user requirements.

Job Scheduling Algorithms

Genetic Algorithm: A method for solving constrained and unconstrained problems of optimization foundation on natural selection, the mechanism that evolved biological process. A population of individual solutions is repeatedly updated by the genetic algorithm. The genetic algorithm selects individuals from the current population at random to be parents at each stage and uses them to produce the babies for the next generation. The genetic algorithm can be used to solve a variety of optimization issues that are not suitable for conventional optimization algorithms where objective function is discontinuous, non-differentiable, stochastic, or highly nonlinear. By using GA, each iteration generates a population of points and selects an optimal solution for the best point in the population. And then, by calculation, select the next population using random number generators.

GA is used in a number of different fields of usage. Hence, by providing arrays of bits or characters to represent the chromosomes, most researchers can introduce this genetic model of computation. Simple operations of bit manipulation allow crossover, mutation and other operations to be performed.

First come first serve (FCFS): FCFS is similar to the data structure of the FIFO (First in First out) queue, where the first data element added to the queue is the first element to exit the queue. In Batch Systems, this is used. Using a queue data structure, where a new process works through the queue's tail, it is easy to programmatically understand and execute, and the scheduler selects the process from the queue end. Purchasing tickets at the ticket counter is a perfect example of FCFS planning in real life. But there is an issue with FCFS, system priority is not taken into account. Parallel use of resources is not feasible and use of bad resources (CPU, I / O, etc.).

Shortest Job First (SJF): A scheduling strategy that chooses the process of waiting to execute next with the smallest execution time. Second, the system is completed, which takes the shortest time to complete the execution. Using ordered FIFO queue, this rule can be enforced. All processes in a queue are sorted on the basis of their CPU bursts. When the CPU is available, a system will be selected to run from the first position in a queue [16].

Round - robin (RR): When the process can be executed, procedures are given an equal slice of time. The execution is carried out one after another on a circular order. So every work has a quantity when it can be done. If this quantity is not sufficient to complete the execution of the process, it will be stopped and the next process will be performed. Upon completing a full round, its turn will come again and so on. If a process is completed, it will go off the list and if another arrives, it will be put at the end of the list waiting for its turn.

There is no hunger in this algorithm, but it can often be too long. This is a typical and conventional load balancing algorithm, but the challenge in round robin is setting the time quantity. To calculate the dynamic quantity of time, various optimization algorithms can be combined with RR [4].

Priority scheduling: It gives a well-defined priority to each system. This way, each process has its own priority, depending on whether it will be run or wait. The first to run is going to be the highest priority operation, while others are going to wait for their turn. First will be handled high-priority activities [20]. Therefore, the low priority tasks would have to wait a long time in the queue. If the system crashes, all non-performing low-priority tasks will be lost.

Min – Min: This algorithm is based on the resource assignment principle that has the shortest completion time (fastest resource), a minimum completion time (MCT) functioned [8]. It is a sample algorithm but it gives the quick result when the size of the task in the task group that deals with other tasks is small compared to the large size task, on the other hand, when large size tasks are performed, it gives poor use of the resource and large maximum completion time of the task since larger tasks have to wait for smaller tasks to be completed. For all tasks to be performed, the Min - Min algorithm first finds the minimum time. Then, it selects the task with the least execution time among all tasks. The algorithm proceeds by assigning the task to the resource generating the minimum completion time. Min - Min repeats the same procedure until all tasks are planned. Min - Min algorithm's limitation is that it first selects smaller tasks that make use of high computational power resource. As a result, when the number of smaller tasks exceeds the large ones, Min-Min's schedule is not optimal.

Max – Min: This algorithm is bottomed on the concept of assigning to the asset, which has the maximum completion time (fastest resource), a task with maximum completion time (MCT). It is a test algorithm but it gives the fast result when the size of the task in the metatask is big compared to the small

size task. On the other hand, if small size tasks are performed, it gives poor use of the resource and a large maximum completion time of the task as smaller tasks have to wait for larger tasks to be completed. The purpose of this algorithm is to prioritize tasks with optimum completion time by executing them first before assigning other tasks with minimum completion time [13].

F. Components of HPC Scheduling

High-performance computing has five components: CPUs, memory, nodes, internodes in the network, and storage (disks, tape). Single-core CPU (processors) is no longer being used today. To date, all CPUs (processors) consist of the configuration used on the motherboard (multiple' cores' on a single' chip'). For a number of reasons, the trend of even more ' core ' per unit will rise. The node plays a major role in linking CPUs, memory, interfaces, computers and other nodes in a physical way. For a high-performance computing system, shared memory is often necessary.

There are five different node types: user node, control node, node of management, node of processing and node of computation, in Fig 2. The client node is the only portal to reach the cluster network for outsiders. Users typically have to log in to compile and run the tasks from the node. Fault-tolerant architecture is accomplished with hardware redundancy to be built in the system to ensure the client node is highly accessible. Control node is mainly responsible for delivering computer node to basic network services such as DHCP (Dynamic Host Control Protocol), DNS (Domain Name Service), NFS (Network File Service) and the distribution of computer node tasks.

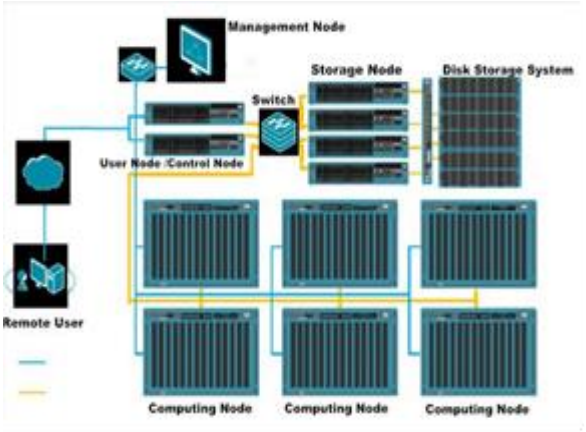

Figure 2. A High Performance Computing Cluster

III. CHALLENGES

In recent study, as the chosen platform for execution as the optimal medium for a wide range of workloads to be performed, there are four major challenges for heterogeneous computing clusters to address.

First, the majority of business applications currently in use were not designed to run on such large, open and heterogeneous computing clusters. To move these applications to heterogeneous computer clusters, particularly with significant performance improvements or improvements in energy efficiency, is a quest and attempt to solve problem. Second, it is also overwhelming to generate new, ground-up enterprise applications to run on the different, heterogeneous computing framework. It is extremely challenging to write high-performance, energy-efficient programs [12]for these architectures to die on an unprecedented scale of parallelism, as well as difficulty in scheduling, interconnecting and sorting. Third, cost savings for utilization and distribution of IT services from the new shared network architecture are only feasible if multiple business applications can share resources in the heterogeneous computing cluster. Nonetheless, allowing multi-tenancy without adversely affecting that application's stringent performance service metrics needs complex scalability and virtualization of a wide variety of different computing, processing and interconnect devices, and this is yet another unresolved issue. Fourth, enterprise systems with unusually heavy load spikes experience greatly variable user loads. Meeting service metrics quality across multiple loads includes the use of an application's elastic computing resources in response to increasing user demand. There are no good solutions to this problem at the moment. There are no good solutions to this obstacle at the moment. The resource management dynamics in HPC are evolving. New application innovations and technical transitions introduce the scheduling models and process architecture with new concepts and specifications.

Another problem is that multicore chips are relatively simple processor core and will be underused if user programs are unable to provide enough parallel thread speed. It is the responsible of the programmer to write parallel high-performance software to make maximum use of the processor core. The new parallel multicore technology should have two characteristics to achieve high performance:

Fine grain thread: a high degree of parallelism is needed to keep each core processor busy. Another

factor is that a core often has to operate on a small-size cache or scratch buffer that allows developers to break down a task into smaller tasks.

Asynchronous program execution: The existence of a synchronization point can seriously affect the performance of the program when there are many processor cores. And reducing unnecessary points of synchronization can subsequently increase the degree of parallelism.

IV. CONCLUSION

We present HPC framework, its scheduling and research challenges in this paper as express in table 2. Task scheduling on parallel machines is a research field that has been well explored and has led to widespread use and there are several methods, including automatic calculations of runtime, partial executions, and smarter allocation schemes for processors. Parallel processing builds on existing technologies implemented in new scenarios and serving diverse users and priorities. Max-min algorithms is the execution of tasks with maximum completion time may first increase the total response time of the system and thus make it inefficient and it has these drawbacks so that researchers may consider the different methods to implement this algorithm. Min-min algorithm is first considered to be the shortest jobs, so it fails to efficiently leverage the resources that lead to a load imbalance, so efficient methods are needed to solve this problem. FCFS which can be a long waiting time if the short request is waiting for the long process [15]. It is not ideal for a system of time sharing where it is critical that each user receives the CPU for the same time interval.

The through in round robin largely depends on the selection of the size of the time quantum. If time quantum is too large it behaves as FCFS. If time quantum is too short much of the time is spent in process switching and hence low throughput and one cannot assign priority to any process which can be a drawback. Priority scheduling will have to consider which parameters are set to high priority within system.

Genetic algorithm has different parameters (size of population, times, rates of mutation, etc.) and there are many problems not well defined (parameters are not well defined or not known).Researchers are need to decide which optimization method to use for research environment to solve problem. Longer processes in shortest job first algorithm have more waiting time, so starvation occurred and need to solve this issue. Some are used average waiting time to reduce long waiting time. We agree that several areas of HPC still need the attention of the researcher. We will have a detailed study of HPC problems and challenges in the future.

TABLE II. COMPARISON OF EACH METHOD

Algorithm	Comparison	Advantages	Disadvantages
Genetic Algorithm (GA)[18]	It is necessary to use as a tool to solve the problems of optimization and search. GA includes several operations to execute the algorithm with several techniques.	This combines good solutions and optimizes towards a goal over time.	Without through too much time for GA to check for optimum results, some new genetic values should be incorporated into the population.
First Come First Serve (FCFS)[15]	FCFS starts imply the first task to be done.	For long process, FCFS is suitable than others and one of the easiest methods.	Each and every small process should wait for its turn to utilize the CPU also, throughput is not emphasized.
Shortest Job First (SJF)[16]	Shortest job starts to be process.	It is optimized for average waiting time.	There is more number of short jobs in a system, the long jobs will be starvation.
Round-Robin (RR) [10]	Time is to be allocated to resources in a time slice manner in this scheduling algorithm.	In a general-purpose, time-sharing or transaction processing system, RR is successful. There is also a small overhead on the processor.	Care must be taken when determining the quantity value, and if the quantity time is too high, the result is also weak.
Priority scheduling	A well-defined priority to each system.	A handled high-priority activity, so user defines process is more suitable.	The low priority tasks would have to wait a long time in the queue of the system.
Min-min [6]	Chooses smaller tasks first to be done.	One resource can execute only one at a time and resources are known in prior.	This is caused load imbalance between large task and small task to execute in the system.
Max-min [6]	Chooses bigger tasks to be completed first.	Prioritize tasks with optimum completion time by executing them first	Execution of task with maximum completion time first might increase the total

		before assigning other tasks with minimum completion time.	response time of the system thus making it inefficient.

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Cyber Security and Digital Forensics

Comparative Analysis of Site-to-Site Layer 2 Virtual Private Networks

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Abstract

Nowadays, many companies have branch offices and connect those offices to the main office over the Internet using a site-to-site Virtual Private Network connection. Most of these connections have always operated at Layer 3 of the OSI network model. In recent years, there has been a growing requirement to extend links at Layer 2, which allows broadcast traffic to be forwarded between sites. Depending on inter-site connection medium, different technologies are utilized. This paper compares and analyses site-to-site Layer 2 VPN technologies, which include layer 2 tunneling protocol (L2TP), and point to point tunneling protocol (PPTP), OpenVPN, Ethernet over IP (EoIP), and MPLS/VPLS to choose the right VPN for the organization. This is done by means of performance measurement and packet analysis. In order to provide fair comparable results, all technologies are tested in the same manner.

Keyword: *PPTP, L2TP, OpenVPN, EoIP, VPLS, Virtual Private Network*

I. INTRODUCTION

While VPNs were commonly planned for individual clients, demand is likewise expanding in business. Organizations presently use VPNs to verify their office systems, business PCs, and Internet connection while others use VPNs to remotely access network resources that are not near them geologically. In the course of the most recent couple of years, VPNs have turned strongly to be one of the most well-known and irreplaceable tools for every privacy-conscious consumer. Globally, the Internet handles around 71,131 GB of traffic for each second, including 2,790,265 emails and 73,849 Google searches for every second [1]. All the company's communications and employees searching for business-related information help to make up those numbers. Furthermore, a breach or leak of the business's data transmitted over the Internet could cost people millions. This raises alarms because, according to a Ponemon Institute survey [2], 67% of SMBs admitted to being attacked in 2018. Large businesses with massive investment may be in a better position to deploy a range of IT security solutions; small businesses need to be more vigilant. The most effective way to prevent the data from reaching the wrong hands is through the use of a VPN service that makes the Internet usage fully private and secure. Businesses may also find different reasons for using a VPN. To fill this gap, there are many kinds of VPN technologies such as IPSec, GRE and SSL. Most of them are Layer 3 VPNs, and it fulfills most of the business requirements. With Layer 3 VPNs, exchange emails and accessing internal servers are easy to use and secure. However, it is not possible to use some software for LAN between two sites, although they are connected by Layer 3 VPN, for example, printer sharing, some database protocols, CRMs, and other applications that are developed for LAN specified purpose. If people want to use LAN applications, a single Ethernet segment needs to be constructed.

Imagine the situation of three remote sites; Yangon, Mandalay and MauBin, and every site have an Ethernet switch. It is a big challenge to connect Ethernet network cables between them. To lay the cable between different offices in different cities is expensive as well as time-consuming. The Internet cannot become an alternative to Ethernet because even if both sites are connected to the Internet, two sites do not construct the single Ethernet segment at all. Layer 2 "Site-to-Site VPN" tunnels the Internet and establish a VPN Session between remote sites with full capabilities to transmit any Ethernet frames. Layer 2 VPN has unlimited protocol transparency, which is identical to physical Ethernet segments. Many protocols such as IPv4 (TCP, UDP, ICMP, ESP, GRE), IPv6, PPPoE, RIP, STP, and others can be used on Ethernet. Any legacy and latest protocols can be used within the Layer 2 VPN sessions. Although provider provisioned Layer 2 VPN solutions such as MPLS/VPLS can be purchased from ISPs, most of these services are monthly payment basis, and the price is not cost-effective.

Not only are these restrictions, but also different IP subnets on each site need to be built. A site's IP subnet cannot overlap with other sites. Moreover, a number of subnets have to be managed in order to prevent any other subnets from colliding. Adopting the Layer 3 VPN for creating site-to-site VPN requires special pain to satisfy the demands of legacy VPNs. However, when we use Layer 2 VPN to link up the site-to-site VPN, it is very straightforward and reduces the effort to coop against several troublesome errors which might occur when Layer 3 VPNs are used. Designing and architecting networks with layer 2 VPNs can be as simple as designing traditional Ethernet network topology with hub-and-spoke mode. Connecting VPN Sessions between sites is possible instead of using physical Ethernet network cables.

All kinds of server and inter-client-PC-communication applications will work well, with no difference between inside the same site and beyond the distance. It is the main reason that the decision to carry out a performance comparison of Layer 2 VPNs is made.

In this paper, the impact of Layer 2 VPNs and performance analysis of five different VPNs, namely, PPTP, L2TP, OpenVPN with BCP, EoIP and MPLS/VPLS are discussed and presented. However, this paper does not provide explicit suggestion on which technology is to be preferred. The rest of this paper is organized as follows: Section II presents related work. Section III explains the characteristic of VPN, and Section IV provides the testbed setup. The experiment results are discussed in section V and draw conclusions in section VI.

II. Related Work

Singh and Gupta [3] proposed Multi-phase encryption and payload encryption; it was applied to the data inside the IP packet of the encapsulated tunnel packet. They discussed the traditional security measures of VPN and a whole new approach for VPN security by using a multi-phase encryption technique. I. Kotuliak, P. Rybár, and P. Trúchly [4] analyzed OpenVPN and IPSec based VPN; they compared those technologies based on parameters such as throughput, the response time of each protocol. They chose OpenVPN due to its simplicity and fast and straightforward implementation.

Chawla et al. [5] explained the architecture and protocols of IPSec and SSL VPN technologies, including their advantages and disadvantages for real kinds of applications. Qin et al. [6] studied IPSec and SSL VPN in detail, and the scope of application, security, scalability, and other aspects are analyzed and compared, advantages and inadequacy are summarized. Zhang Zhipeng et al. [7] presented three types of common VPNs and explained a comparative study of their features, performance, security, and cost-efficient.

None of the related works compared to the performance of Layer 2 VPNs. In this paper, we concentrate on the performance of Layer 2 VPNs.

III. Characteristics and Models of VPNs

A plethora of methods is used to model and characterize VPNs. The purpose of this section is to introduce and explain each of these models and characterizations.

A. Service Provider and Customer Provisioned VPNs

VPNs that are configured and managed by a service provider are service provider provisioned VPNs. VPNs that are configured and managed by the customer itself are called customer provisioned VPNs. Examples of service provider provisioned, and customer provisioned VPNs are shown in Table 1.

TABLE 1. Service Provider and Customer Provisioned VPNs

Provider Provisioned	Customer Provisioned
VPWS, VPLS, IPLS	PPTP, L2TP, OpenVPN
BGP/MPLS, IPSec, GRE, IP-in-IP	IPSec, GRE, EoIP

B. Site-To-Site and Remote Access VPNs

Whether provider or customer provisioned, VPNs fall into one of two broad categories: site to site or remote access. Site-to-site VPNs allow connectivity between an organization's geographically dispersed sites (such as a head office and branch offices). Fig 1 illustrates a typical site-to-site VPN.

Remote access VPNs allow mobile or home-based users to access an organization's resources remotely. Fig. 2 illustrates typical remote access VPNs.

C. Protocol Background

This section presents protocols used in Layer 2 VPN technologies.

1)PPTP: The Point to Point Tunneling Protocol (PPTP) is one of the oldest protocol. PPTP uses the TCP port 1723 for remote access over the Internet. The data packets transmitted through the tunnel are

encapsulated. It is suitable for applications where speed is important, such as streaming and gaming.

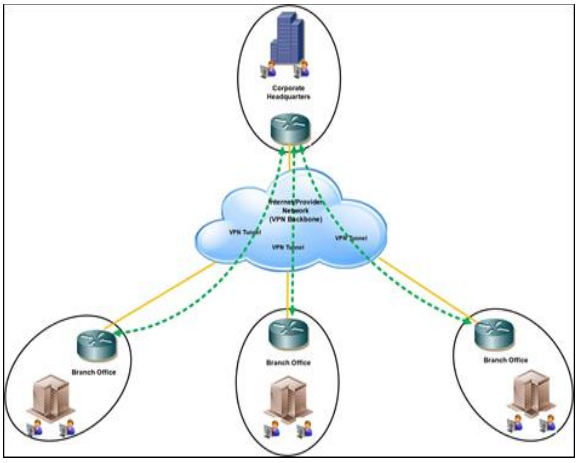

Figure 1. Typical site-to-site VPN

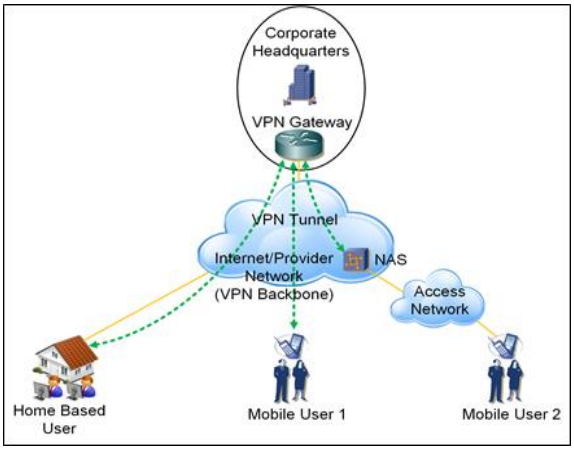

Figure 2. Remote Access VPNs

2)L2TP with IPSec: L2TP stands for Layer 2 Tunneling Protocol and does not provide any encryption on its own. L2TP usually uses IPSec (Internet Protocol Security) authentication protocol. The data transmitted through the L2TP / IPSec protocol is usually authenticated twice. Each data packet transmitted through the tunnel includes L2TP headers. One of the many reasons why L2TP is a common protocol is that there are no known vulnerabilities.

3)OpenVPN: OpenVPN is often referred to as an SSL-based VPN because it uses the SSL/TLS protocol for secure communication. The control channel is encrypted and protected using SSL/TLS while the data channel is encrypted using a custom encryption protocol. OpenVPN's default protocol and port are UDP and port 1194.

4)PPP Bridging Control Protocol: BCP allows bridging the Ethernet frame through the PPP link. Established BCP is an integral part of the PPP tunnel. The Bridging Control Protocol (BCP) is responsible for configuring, activating and disabling the bridge protocol modules at both ends of the point-to-point link. PPTP, L2TP, and OpenVPN protocols can carry only the upper layer of Layer 3 and more. However, with the support of BCP, they can work as Layer 2.

5)EoIP with IPSec: IP protocol 47/GRE allows tunnel creation by encapsulating Ethernet frames in IP packets and forwarding them to another router. Ethernet over IP (EoIP) establishes an Ethernet tunnel on top of an IP connection between two routers. All Ethernet traffic will be bridged, just as if there is a physical interface.

6)MPLS VPLS: Virtual Private LAN Service (VPLS) offers multipoint Ethernet-based connectivity over IP or MPLS networks. It enables geographically dispersed sites to share an Ethernet broadcast domain by linking sites through pseudowires. It is often used for extending LAN services over a network given by a service provider.

IV. Testbed Setup and Performance Parameters

This section describes how to setup testbed to measure performance and to analyze security.

A. Testbed Setup

There are two laptop computers and three desktop computers in this setup. WAN Emulator [8] is running on a desktop computer. RouterOS [9] is running on two computers to create a tunnel between these two desktop computers. Two laptops are running iPerf software to test throughput. In testbed example, the iPerf client send the 100MB of data to the iPerf server, and the output are saved in CSV file. Our testbed setup is as shown in Fig. 3 and their hardware and software specifications are shown in Table 2.

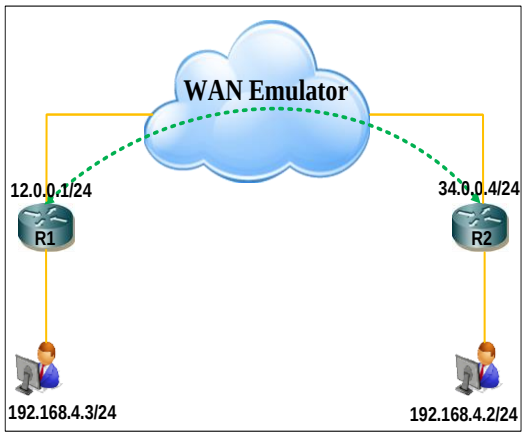

Figure 3. Testbed setup

TABLE 2. Hardware and Software Specification

Type	Description
Laptop x 2	Intel(R) Core(TM) i5-7200 CPU @ 2.50GHz(4CPUs), ~2.7GHz, 8GB memory
Desktop x 3	Intel(R) Core(TM) i3-2100 CPU @ 3.10GHz(4CPUs)
WANem	Wide Area Network Emulator v3.0 Beta 2 released
VirtualBox [10]	Oracle VirtualBox 6.0 hypervisor software
RouterOS	6.46.1 (Stable) release

In this paper, WANEM is used to emulate to define QoS parameters such as packet loss, jitter, and delay. Packet loss has a direct impact on the stability of the VPN. Packet loss occurs when the network is congested.

TABLE 3. QoS Parameters

Parameters	Value
Bandwidth Limit	50 Mbps
Delay	20ms, 40ms, 60ms
Jitter	2ms
Packet Loss	0.1%

Delay is the amount of time a packet travels from its source to destination. Jitter is the changing rate of delay across a network, and is measured in milliseconds and it has a great impact on live streaming application such as video and VoIP. To be similar with real network, QoS values are defined as shown in Table 3.

B. Measurement Tools

Assessing the performance of Layer 2 VPN requires the use of several measurement tools for generating, measuring, and monitoring network traffic. The tools used in this work are Wireshark [11] and iPerf3 [12]. Wireshark is a network protocol analyzer with a rich feature set for capturing and analyzing network traffic. iPerf3 is a network testing tool for active measurements of the maximum achievable bandwidth on IP networks.

V. Experimental Results and Discussion

Experimental results are based on the different parameters for the different VPN technologies. This section shows the performance of seven different methods of Layer 2 VPNs in terms of throughput and protocol analysis.

A. Throughput

Throughput is measured in bits per second. By analyzing the results, throughput varies depending on protocol nature and encryption method. For throughput measurement, iPerf3 is used to exchange traffic between two laptops. For all VPN technologies, the same amount of traffic (100MB) is exchanged and tested three times with three different delays; 20 milliseconds (ms), 40 ms, and 60 ms. The results are documented in Table 4, and Fig. 4 shows the throughput comparisons.

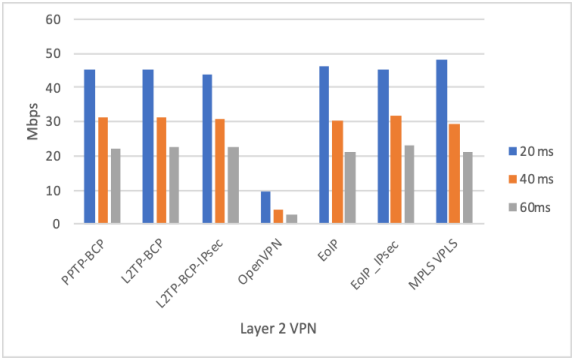

Figure 4. Throughput comparison

TABLE 4. Throughput Comparison with Different Delays

VPN	Delay 20 ms	Delay 40 ms	Delay 60 ms
PPTP-BCP	45.1 Mbps	31.4 Mbps	22.2 Mbps
L2TP-BCP	45.2 Mbps	31.4 Mbps	22.6 Mbps
L2TP-BCP-IPSec	44 Mbps	30.8 Mbps	22.7 Mbps
OpenVPN	9.6 Mbps	4.16 Mbps	2.6 Mbps
EoIP	46 Mbps	30.4 Mbps	21.2 Mbps
EoIP_IPSec	45.1 Mbps	31.8 Mbps	23.1 Mbps
MPLS VPLS	48.1 Mbps	29.4 Mbps	21.3 Mbps

TABLE 5. Throughput and Loss Comparison between Non-VPN Traffic and VPN Traffic at 20ms delay

VPN	Non-VPN Throughput	VPN Throughput	% of Loss
PPTP-BCP	50 Mbps	45.1 Mbps	9.8%
L2TP-BCP	50 Mbps	45.2 Mbps	9.6%
L2TP-BCP-IPSec	50 Mbps	44 Mbps	12.0%
OpenVPN	50 Mbps	9.6 Mbps	80.8%
EoIP	50 Mbps	46 Mbps	8.0%
EoIP_IPSec	50 Mbps	45.1 Mbps	9.8%
MPLS VPLS	50 Mbps	48.1 Mbps	3.8%

All VPN tunnels can degrade performance because of the overhead and encryption methods they use. The amount of throughput loss is due to the trade-off between network performance and security. Generally, the more secure tunnel may result in poor throughput, while less secure tunnel may have better throughput. Table 5 shows the loss of throughput when traversing a tunnel.

B. Packet Analysis

Wireshark protocol analyzer captures the traffic and analyze while two computers ping each other inside Layer 2 VPN tunnels.

1)PPTP with BCP: Fig. 5 shows the packet analysis for the PPTP with BCP, and it can be seen clearly that two computers ping each other since there is no encryption with PPTP.

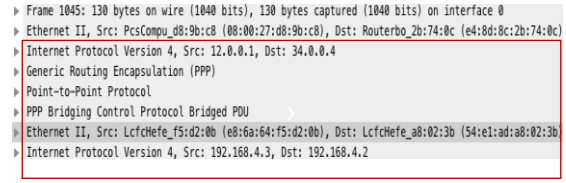

Figure 5. PPTP with BCP

2) L2TP with BCP: When analyzing the packets transmitted through L2TP with BCP tunnel, it is observed that which protocols, along with which source addresses and destination addresses that are being used can be sniffed.

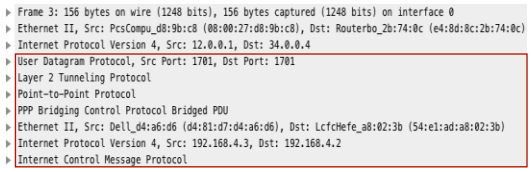

Figure 6. L2TP with BCP

3) L2TP with IPSec with BCP: When analyzing the packets transmitted through L2TP with BCP tunnel with IPSec encryption, only the information of encapsulated payload can be sniffed.

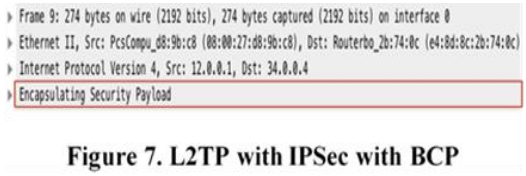

Figure 7. L2TP with IPSec with BCP

4) OpenVPN: Due to its secure encryption methods used, packets that are transmitted through OpenVPN display only the OpenVPN protocol with no other additional information.

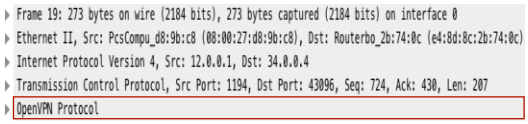

Figure 8. OpenVPN

5) EoIP: Similar to unencrypted L2TP and PPTP tunnels, EoIP also shows protocol and addresses of both source and destination when analyzed by a packet sniffer.

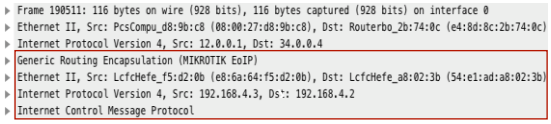

Figure 9. EoIP

6) EoIP with IPSec: EoIP with IPSec no longer displays which protocols along with source and destination addresses accessed in packet analysis if it is properly encrypted by IPSec.

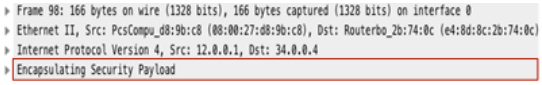

Figure 10. EoIP with IPSec

7) MPLS VPLS: When examine the packets transmitted with MPLS VPLS, it is observed which protocols can be sniffed along with the source addresses and destination addresses used.

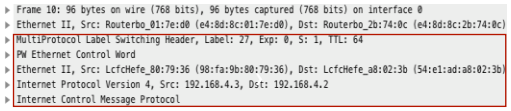

Figure 11. MPLS VPLS

TABLE 6. Comparison Matrix of Authentication and Encryption

VPN Type	Encryption	Authentication	Can be bridge
PPTP	MPPE128	Username Password	With BCP
L2TP	IPSec	Username Password	With BCP
OpenVPN	TLS (AES/BF)	TLS	With BCP
EoIP	IPSec	No	Yes
MPLS/ VPLS	No	No	Pseudowires & Control Word

C. VPN Selection

This section discusses the use of each VPN based on the various throughput performance test and packet analysis conducted in previous sections. Packet analysis describes that OpenVPN, L2TP IPSec with

BCP, and EoIP with IPSec are good for security. The throughput result shows that MPLS VPLS is 48.1 Mbps at 20 ms. EoIP with IPSec is 31.8 Mbps at 40 ms, and 23.1 Mbps at 60 ms. Although the result of EoIP is good at 40 ms and 60 ms, it is not widely used in the industry because it is not mature yet and still vendor dependent. As mentioned in section III, MPLS/VPLS is a provider provisioned VPN, and customer cannot manage themselves. L2TP IPSec with BCP should be considered in term of performance and security perspective for enterprise networks which need Layer 2 VPN connections. The pros and cons of each VPN on various aspects can be observed in Table 7.

TABLE 7. Pros and Cons of Different VPNs

	Security	QoS	Scalability	Cost
PPTP-BCP	Low	No	Good	Low
L2TP-BCP	Low	No	Good	Low
L2TP-BCP-IPSec	High	No	Good	Average
OpenVPN	Higher	No	Good	Average
EoIP	Low	Yes	Average	High
EoIP_IPSec	High	Yes	Average	High
MPLS VPLS	Average	Yes	Best	Higher

VI. CONCLUSION

The main purpose of this paper is to analyze and compare site-to-site Layer 2 VPNs. The experimental results are achieved with different throughputs from five different VPN technologies. They are monitored and captured by Wireshark network protocol analyzer so as to see what protocols and overheads are added to the original frame inside layer 2 tunnel. It is easy to see that Layer 2 VPN carries Ethernet frame that can raise the overhead compared to Layer 3 VPN. As a result of this study, it is not easy to recommend one VPN against to the other because each one of them has advantages and disadvantages in term of security and performance. VPN protocol's encryption capabilities are paramount important because it determines the level of privacy and protection, however, this should not be only one reason to choose the VPN for organization. Therefore, organization should consider VPN technology that can balance performance as well as security.

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Effect of Stabilization Control for Cooperation between Tele-Robot Systems with Force Feedback by Using Master-Slave Relation

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Abstract

By experiment, this paper investigates the effect of the stabilization control with filters to suppress instability phenomena for tele-robot systems with force feedback by using a master-slave relation. As the quality of service (QoS) control to improve the quality of experience (QoE), the systems carry out the adaptive Δ-causality control, which we previously proposed. In our experiment, we carry a wooden stick together by gripping the two ends of the stick with the two robot arms of the systems. Experimental results illustrate that the stabilization control with filters can suppress instability phenomena.

Keywords—Tele-Robot System, Remote Control, Force Feedback, Cooperation, Stabilization Control, QoS, Experiment

I. INTRODUCTION

By using multiple bilateral tele-robot (i.e., remote robot) systems with force feedback [1]-[5], we can conduct cooperative work among the systems [6], [7]. This is because a user of such a system can operate a tele-industrial robot having a force sensor via a network by employing a haptic interface device while watching video of the robot motion. By using force feedback for the robot operation, the user can feel the reaction force outputted by the haptic interface device, and he/she can perform more accurate operation. However, if the information about force is transmitted via a network without the quality of service (QoS) [8] guarantee such as the Internet, the quality of experience (QoE) [9] may seriously be degraded and the instability phenomena may occur owing to the network delay, delay jitter, and so on. In order to solve the problems, we need to perform QoS control [8] and stabilization control [10] together.

In [6], the authors investigate the influence of the network delay on cooperative work in which a single user operates the haptic interface devices of the two tele-robot systems with his/her both hands, and carries an object held by the two robot arms. The systems have an equal relationship in this paper, but a master-slave relation between the systems is also important. In [7], thus, the authors conduct the same work as that in [6] by using the tele-robot systems with a master-slave relation and apply the adaptive Δ-causality control [11] to avoid large force applied to the object. As a result, it is illustrated that the control can suppress large force even if the network delay becomes larger. However, to avoid instability phenomena, the reaction force outputted from the haptic interface device is set to a small value by multiplying 0.5 to the force detected by the robot's force sensor. We need to perceive larger reaction force to conduct the work more precisely. To solve the problem, we need to perform the stabilization control with filters [10]. However, the effect of the control has not been clarified quantitatively in the systems using the master-slave relation.

Therefore, in this paper, we perform the stabilization control with filters for the tele-robot systems using the master-slave relation with force feedback in which the adaptive Δ-causality control is exerted. By experiment, we clarify the effect of the stabilization control. We also investigate the effect of the adaptive Δ-causality control under the stabilization control.

The remainder of this paper is organized as follows. In Section II, first we give an outline of the tele-robot systems with force feedback by using the master-slave relation. Next, we explain the adaptive Δ-causality control in Section III, the stabilization control with filters in Section IV, and experiment method in Section V. Then, Section VI presents experimental results and discusses them. Finally, Section VII concludes the paper.

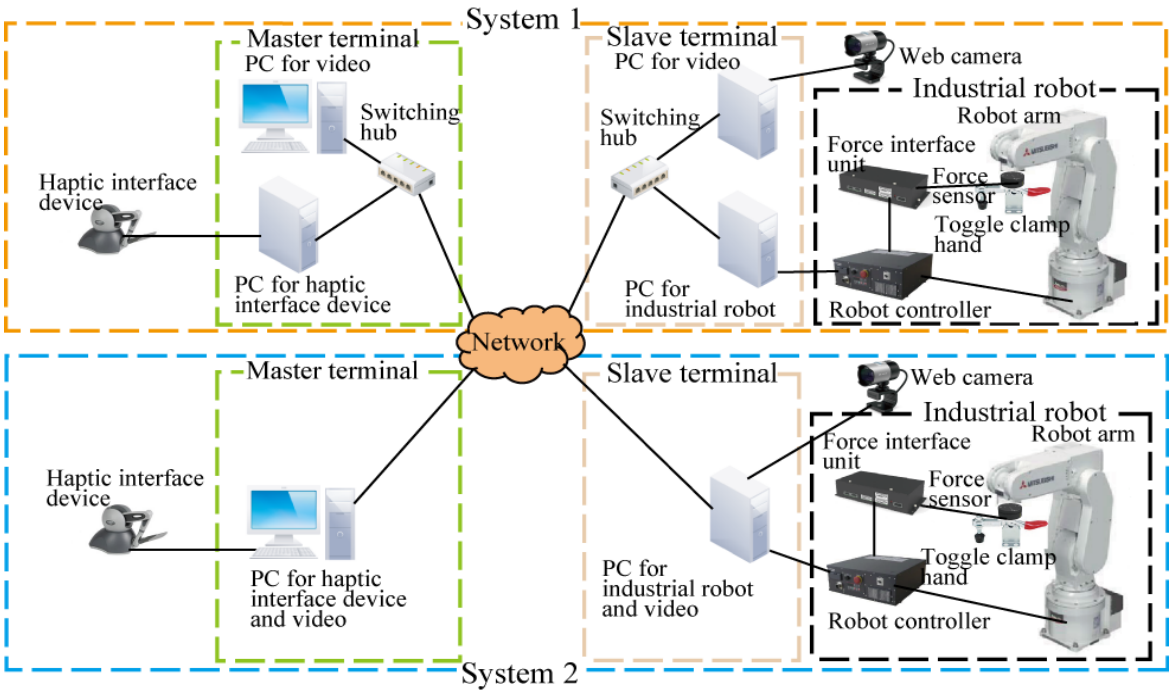


Figure 1. Configuration of tele-robot systems with force feedback.

II. Tele-Robot Systems with Force Feedback by Using Master-Slave Relation

In this section, we outline the tele-robot systems with force feedback in Subsection II.*A* and how to calculate the reaction force in Subsection II.*B*. We also describe the tele-robot systems by using the master-slave relation in Subsection II.*C*.

A. Tele-Robot Systems with Force Feedback

As shown in Figure 1, each of the two systems (called *systems 1 and 2* here) consists of a master terminal with a haptic interface device (3D Systems Touch [12]), and a slave terminal with an industrial robot and a web camera (made by Microsoft). The master terminal in system 1 is composed of two PCs; one is for haptic interface device, and the other is for video. The two PCs are connected by a switching hub. One PC called PC for haptic interface device and video is used in system 2, and the functions of PC are same as those in system 1. The slave terminal in system 1 also consists of two PCs (called PC for industrial robot and PC for video) which are connected by a switching hub. One PC (called PC for haptic interface device and video) is employed in system 2. PC for industrial robot (or PC for industrial robot and video) is connected to the industrial robot directly by a 100BASE-TX cable. The industrial robot consists of a robot controller (CR750-Q), a force interface unit (2F-TZ561), and a robot arm with 6 DoF (Degree of Freedom) (RV-2FB-Q). The force sensor (1F-FS001-W200) is linked to the tip of the robot arm. Furthermore, a toggle clamp hand is attached to the force sensor. We use the toggle clamp hand to fix an object by a toggle. The reaction force outputted by the haptic interface device is calculated from the value obtained by the sensor (the calculation method will be described in the next subsection).

B. Calculation Method of Reaction Force

The reaction force $\boldsymbol{F}_t^{(\mathrm{m})}$ outputted by the haptic interface device at the master terminal at time t (≥ 1) is calculated by the following equation:

$$\boldsymbol{F}_t^{(\mathrm{m})} = K_{\mathrm{scale}} \boldsymbol{F}_{t-1}^{(\mathrm{s})} \tag{1}$$

where $\boldsymbol{F}_t^{(\mathrm{s})}$ is the force obtained from the slave terminal at time t, and K_{scale} (> 0) is the scaling factor applied to $\boldsymbol{F}_{t-1}^{(\mathrm{s})}$. In this paper, we set K_{scale} = 1.0 (note that K_{scale}= 0.5 in [7]).

Furthermore, the position vector $\boldsymbol{S}_t$ of the industrial robot at time t (≥ 2) can be obtained from the following equation:

$$\boldsymbol{S}_t = \begin{cases} \boldsymbol{M}_{t-1} + \boldsymbol{V}_{t-1} \ (\text{if } |\boldsymbol{V}_{t-1}| \geq V_{\max}) \\ \boldsymbol{M}_{t-1} + V_{\max} \dfrac{\boldsymbol{V}_{t-1}}{|\boldsymbol{V}_{t-1}|} \ (\text{otherwise}) \end{cases} \tag{2}$$

where $\boldsymbol{M}_t$ is the position vector of the haptic interface device that the slave terminal receives from the master terminal at time t, and $\boldsymbol{V}_t$ is the velocity

vector of the robot arm at time t. $V_{\max}$ is the maximum moving velocity, and the moving amount is limited so that the robot does not move too fast. In this paper, we set $V_{\max} = 5$ mm / s [13].

C. *Master-Slave Relation*

The two tele-robot systems in Subsection II.*A* are used as follows. One system is master, and the other is slave. The robot of the slave system (called the slave robot here) works according to the movement of the robot of the master system (called the master robot). Position information of the master robot is transmitted from the slave terminal of the master system to the slave terminal of the slave system, and the slave robot is controlled by using the information (i.e., unilateral control). For simplicity, the position information of the haptic interface device of the slave system is not transmitted to the slave terminal. A user of the master system operates the haptic interface device of the master system, and another user of the slave system can feel the reaction force by holding the haptic interface device of the slave system.

III. Adaptive Δ-Causality Control

When the network delay between the master and slave systems is large, the slave robot lags behind the master robot in the movement of robot arm. Then, the force applied to an object carried by the two robot arms becomes larger, and the operability of the haptic interface device is significantly degraded. We reduce the force and improve the operability by carrying out the adaptive Δ-causality control [10], which delays the output timing of the robot's position information dynamically according to the network delay; by this, because we can delay the master robot's operation by the network delay from the slave terminal of the master system to the slave terminal of the slave system, the operations at both robots are performed at almost the same time.

The adaptive Δ-causality control outputs position information at the generation time (i.e., the timestamp) + Δ (> 0) seconds if the information is received by the time + Δ. Otherwise, the information is discarded as old and useless information to keep the causality. The minimum value Δ_L and the maximum value Δ_H ($\Delta_H \geq \Delta_L > 0$) are set for Δ. Also, since Δ

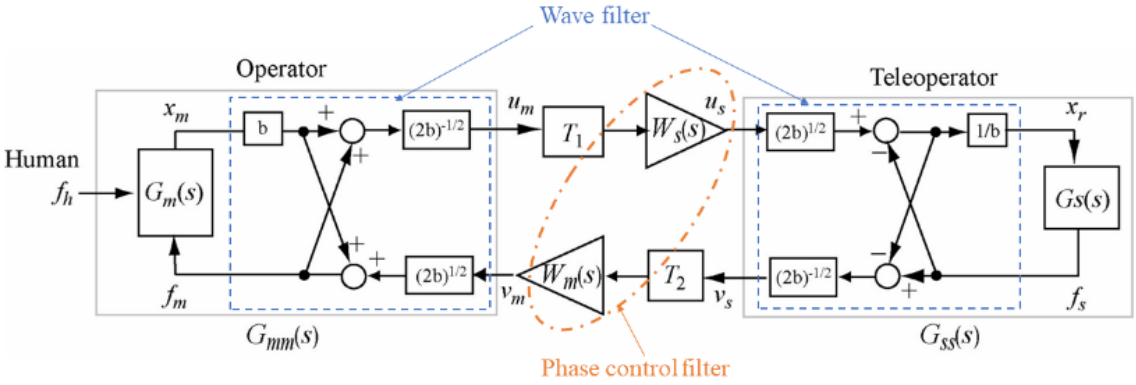
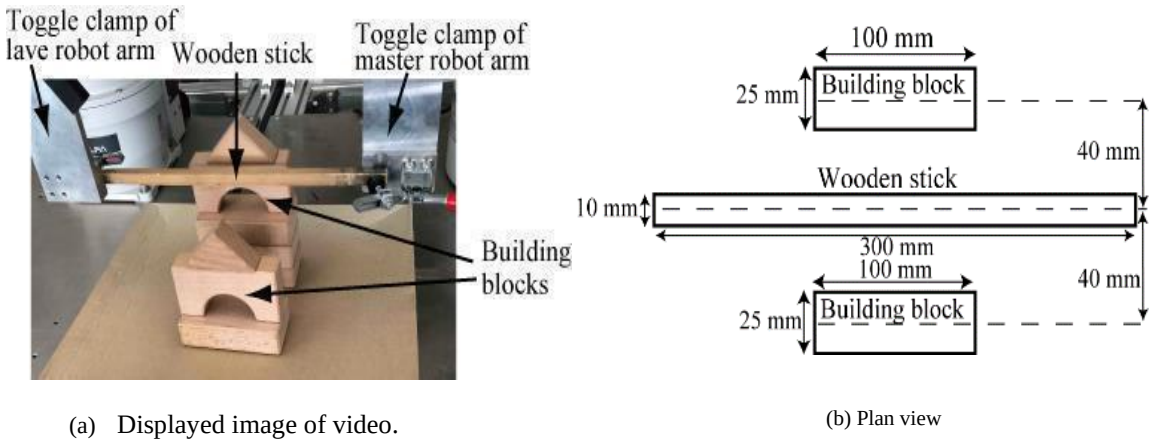


Figure 2. Block diagram of stabilization control with filter.



(a) Displayed image of video.

(b) Plan view

Figure 3. Positional relationships between wooden stick and building blocks.

changes dynamically according to the network delay, the value D_t ($t \geq 0$) obtained by smoothing the network delay d_t measured at time t by the following equation is used as Δ:

$$\begin{cases} D_0 = d_0 \\ D_t = \alpha D_{t-1} + (1 - \alpha) d_t \ (t \geq 1) \end{cases} \quad (3)$$

where α is a smoothing coefficient, and we set α = 0.998 [15] in this paper.

IV. Stabilization Control

This section gives an outline of the stabilization control with filters. Figure 2 shows the block diagram of the stabilization control. The control employs the wave filter in combination with the phase control filter [10], [15]. It can make the tele-robot systems with force feedback stable for any network delay. The reader is referred to [10], [15], and [16] for details of the control.

V. Experiment Method

In this paper, for simplicity, the experiment was performed by one person, and the video was watched only at the master system. In the experimental system, the master and slave systems were connected via a network emulator (NIST Net [17]), and a constant delay was added to each packet transferred in both directions between the two slave terminals (the one-way constant delay is called the additional delay here). The additional delay between the master and slave terminals in each system was 0 ms for simplicity.

In the experiment, in order to move the robot arms in almost the same way always, we carried out work of pushing the top block piled up front and behind the initial position of the wooden stick held by the two toggle clamp hands of the robot arms for about 15 seconds (it took about 5 seconds to drop the front block, and around 10 seconds to drop the behind block. Note that the hands change the movement directions after dropping the front block). Figure 3 shows the positional relationships between the wooden stick and building blocks; the position difference between the front and behind brocks is 80 mm. The height of the behind block is 50 mm higher than the front block as shown in Figure 3 (a). In order to realize more stable operation, the motion of the robot arm in the left and right (the *y*-axis) direction was stopped, and the motion was performed only in the front and behind (the *x*-axis) and up and down (the *z*-axis) directions [6].

We changed the additional delay between the two robots of the master and slave systems to 0 ms and 200 ms with and without the adaptive Δ-causality control (called *control* and *no control*, respectively, in Section VI). Then, we measured the position and the force detected by the force sensor.

VI. Experimental Results

We show the position and force of each robot's x and z axes as a function of the elapsed time from the beginning of the work in Figures 4 and 5. We set the additional delay to 200 ms in the figures.

In Figures 4 and 5, we see that there is no instability phenomenon in the tele-robot systems even though the reaction force output from the device is multiplied by 1.0 to the force detected by the robot's force sensor (K_{scale} = 1.0). However, instability phenomena (i.e., large vibrations) of the robot arms occurred and we could not carry out the work when we did not carry out the stabilization control; even if the additional delay is 0 ms, the phenomena occurred. Thus, we do not show the position and force in this case. On the other hand, we confirmed that there was no instability phenomenon under the stabilization control with filters when the additional delay is less than or equal to at least 400 ms; we obtained almost the same results as those in Figures 4 and 5. Therefore, we can say that the instability phenomena of the robot arms are greatly suppressed by the stabilization control.

We find in Figures 4 and 5 that the position and force of no control fluctuate greatly, but those of control are suppressed. In Figures 4 (a) and (c), we observe that the position of the salve robot is about 200 ms behind that of the master robot. However, we confirm in Figures 5 (a) and (c) that the two positions are almost the same. These are the effects of the adaptive Δ-causality control under the stabilization control.

Furthermore, from Figures 4 and 5, we can confirm that the movement direction of the robot is reversed at about 6 second, and the sign of the force is also reversed. This is because the direction of movement to drop the behind block after dropping the front block is changed at about 6 second. Figures 4 and 5 reveal that the force in *x*-axis is larger than the force in *z*-axis under control, but the magnitudes of force in the x-axis and *z*-axis directions are almost the same under control.

VII. Conclusion

In this paper, we applied the stabilization control with filters for cooperative work between the tele-robot systems with force feedback by using a master-slave relation. As QoS control, we carried out the adaptive Δ-causality control, and we also investigated the effect of

the stabilization control. As a result, we found that the instability phenomena can greatly be suppressed by the control in the systems. We also saw that the adaptive Δ-causality control is effective under the stabilization control.

As our future work, we will apply the stabilization control with filters and the adaptive Δ-causality control to the systems with an equal relationship and investigate their effects. We will also switch the master-slave relationship dynamically according to the network delay in the systems.

ACKNOWLEDGMENT

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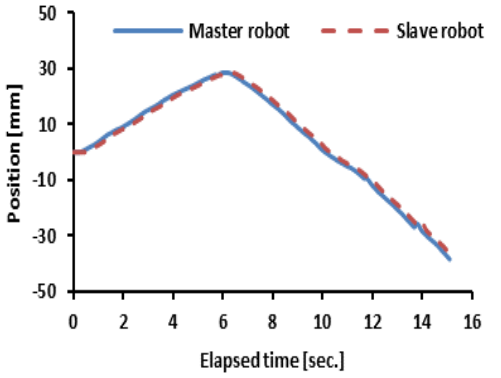
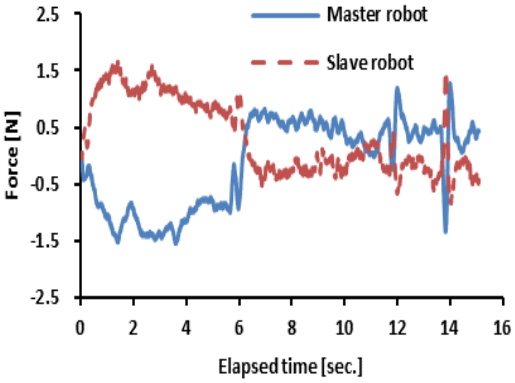
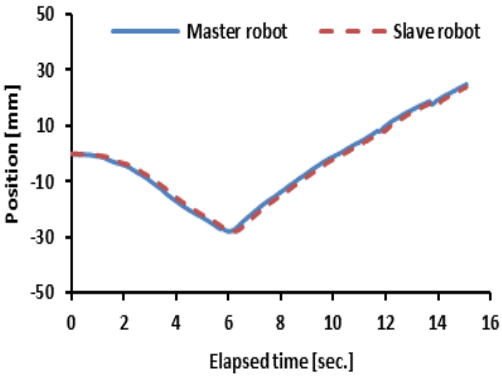


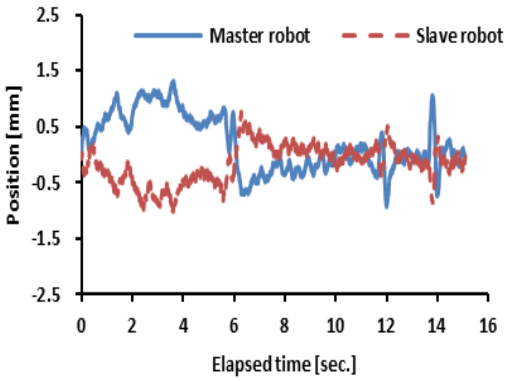
(a) Position of *x*-axis



(b) Force of *x*-axis

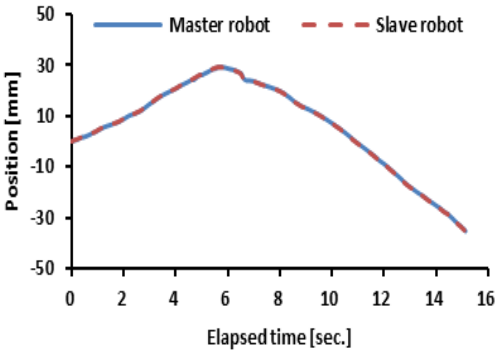


(c) Position of *z*-axis

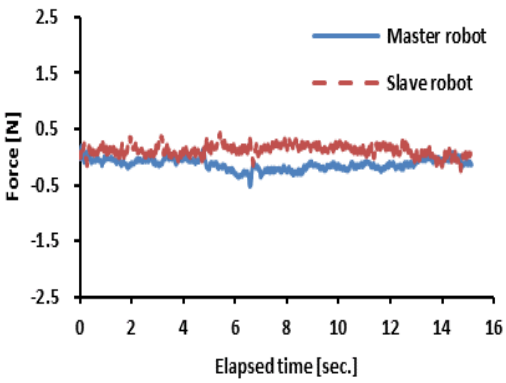


(d) Force of *z*-axis

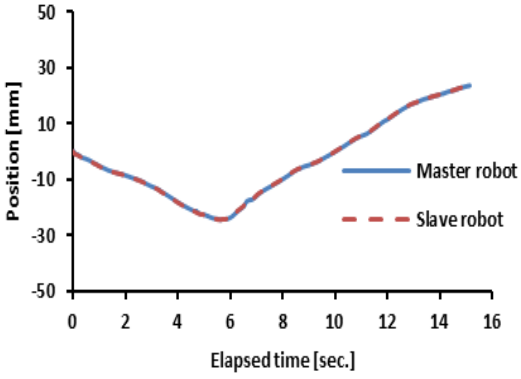
Figure 4. Robot position and force vs. elapsed time under no control (additional delay: 200 ms).



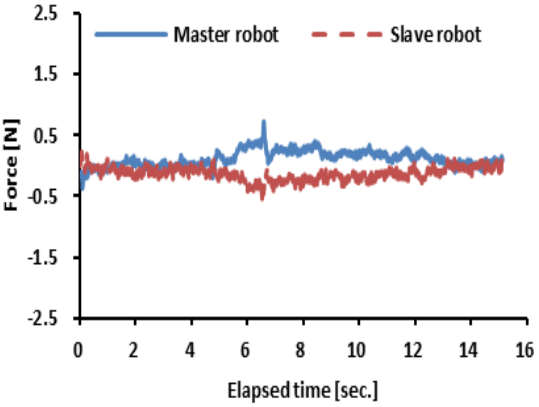
(a) Position of *x*-axis



(b) Force of *x*-axis



(c) Position of *z*-axis



(d) Force of *z*-axis

Figure 5. Robot position and force vs. elapsed time under control (additional delay: 200 ms).

Experimental Design and Analysis of Vehicle Mobility in WiFi Network

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Abstract

Instead of using static nodes in wireless sensor network, implementation of hardware and mobile environment are challenges of mobile wireless sensor network. Mobile sensor nodes combined with wireless network is one of research area and mobility is one of key factors in designing mobile wireless sensor network. To analyze vehicle mobility, this work implements a mobile sensor node in WiFi network and it is equipped with ultrasonic sensor and ESP8266 NodeMCU. The sensor data can be monitored on ThingSpeak over the internet via wireless LAN and its measurement can be taken within 10 s delay. MikroTik (RB931-2nD) router is used as an access point in system architecture. Experiments are performed to study mobility effects on network throughput when mobile sensor node is moving with constant velocity of 10 cm/s. With connectivity in wireless LAN 802.11/n technology, signal strength and throughput are measured for vehicle mobility.

Keywords: *Vehicle Mobility, WiFi, Mobile Wireless Sensor Network, ThingSpeak, Throughput Analysis*

I. INTRODUCTION

Several mobility patterns can be classified as follows; pedestrian, vehicles, aerial, robot and others. Regarding mobility in wireless sensor network with WiFi connectivity, reducing friction and allowing high speed can be obtained in wheel based vehicular mobility. To prevent collisions, all vehicles are planned to move in paths for bounded in 1D movement. In this work, assumption on constant speed and linear movement are made to be bounded in mobility pattern. A mobile sensor node is implemented as parasitic mobility. This mobility pattern can present sensor network performance where sensor node harvests the mobility from vehicle. To measure mobility pattern in wireless sensor network, localization of sensor node can help to analyze the network behavior [1]. In this paper, measuring vehicle mobility is analyzed by using range-based localization technique. In this ultrasonic sensor based for range information, basic principles of ultrasonic range finder are used to be implemented with Arduino. Experimental study is done for analysis of mobility in connectivity of wireless local area network. The rest of paper is organized as follows: related work is described in section 2. Definition and assumption in this paper is mentioned in section 3. Section 4 gives the results and discussion and section 5 gives the conclusions of this work.

II. RELATED WORK

The research work on mobility are studied in literature and presented in sections 2.1, 2.2 and 2.3.

A. Localization in Mobility

Mobility can help to solve the typical problems such as low node density, obstacles and concave topologies. In addition, mobile assisted localization can give better localization accuracy and coverage. Survey of localization problems and solutions are analyzed for wireless sensor network in [2]. In wireless sensor network, range free localization techniques can be used for moving target nodes. In [3], localization algorithm was used to demonstrate localization accuracy and stability. The dynamic aspect of network coverage can be characterized and identified in [4] on the process of sensor movement in optimal mobility strategies.

B. Network Topology in Mobility

Due to the mobility of mobile node, network topology can change significantly for long time scale. Several researches considered mobility effect in Ad Hoc network model. In [5], per-session throughout is kept constant with increasing number of nodes per unit area when users move independently around the network. Investigation of node mobility effect in finding best relay node are proposed in [6] to improve the throughput in Ad Hoc network model. Ad hoc is

one of wireless mesh typology. Rapid growth of IoT applications, mesh typology is attractive for heterogeneous network in which coverage is one of factors [7].

C. Mobility in Wireless Sensor Network

Data drive is the main difference between sensor network and mobile Ad Hoc network. Instead of using static sensor node in network, mobile sensor nodes are necessary in real applications. In [8], mobility, typology and localization are key factors in designing mobile wireless sensor network. The mobility and sharing of internet enabled wireless sensor network are analyzed for specific applications [9]. Hardware cost is one of design challenges for implementing mobile sensor node in wireless sensor network. Nowadays, microcontroller-based sensor nodes are easily implemented with additional unit of localization finder to be implemented a mobile sensor node.

III. Definition and Assumptions

In this section, definition and some models that used in this paper is introduced.

A. Throughput

Throughput of the network is the sum of per-node throughput for all nodes in a network. Per-node throughput of λ(n) bits/s can be defined as the time average of transmitted bits by each node to its destination [10]. In this work, destination is gateway router. In a network, per-node throughput capacity is of order of f(n) bits per second if a deterministic constant $0<c1<+\infty$.

B. Network Model

Assumption in this network is that all n mobile sensor nodes are independently and randomly distributed in defined area. To localize parasitic mobility, generated traffic denoted as λ(n) bits/s at each sensor node is dependent of vehicle mobility process. A mobile sensor node sends a packet directly to fixed router if it is within transmission range of mobile sensor node.

C. Random Direction Mobility Model

In this mobility model, mobile sensor node must travel within defined area at a constant speed and direction. After the nodes pause, a new direction and velocity is chosen randomly and then the process will repeat [11]. The parameters for point-to-point segment *i* is as random variables with the following uniform distributions:

- Absolute angle $\varphi_i = \Phi$: uniform $[0, 2\pi]$
- Distance traveled $l_i = L$: uniform[Lmin, Lmax]
- The speed $v_i = V$: uniform[Vmin, Vmax]
- The pause $t_i = T$: uniform[Tmin, Tmax]

IV. Tests and Results

To summarize the entire system work in Fig. 1, the system comprises three portions: wireless mobile sensor node, fixed router and monitoring server.

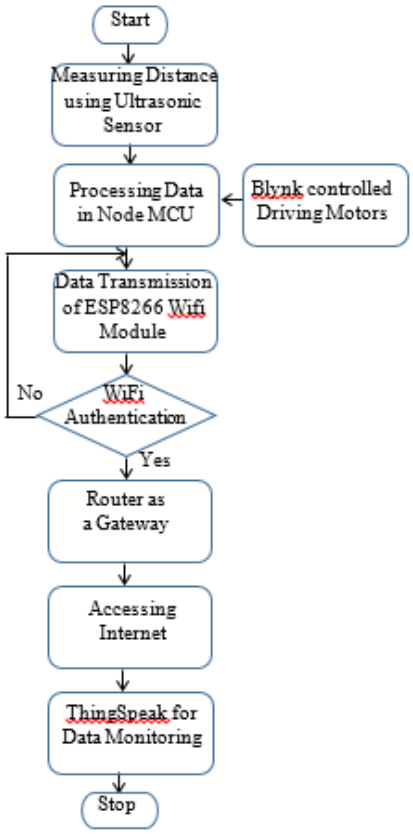

Figure 1. Flow chart for system process

Wireless mobile sensor node senses the distance between it and the facing obstacles by using an ultrasonic sensor. Fixed router receives the packages from mobile sensor node and routes them to the destination. And the ThingSpeak IoT analytic platform is used as a monitoring service provider in this system.

All the router configuration is managed with the use of Winbox software. For local network, IP address of Mikrotik router is set to 172.20.23.17 with a subnet mask of 255.255.255.240 whose prefix is (/28). DHCP

server is configured in the router. For WiFi interface, configuration includes WiFi hotspot creating and setting authentication features for wireless protocol IEEE 802.11. For monitoring sensor data on ThingSpeak, a new channel is created after creating user account. Unique channel ID 774771 is used in testing and the feed data of channel can be viewed with private access or public access. There are two API keys and feed data is imported as a CSV file format. For connectivity of ThingSpeak server and NodeMCU, ThingSpeak library is installed in the Aurdino IDE software.

A. System Description

The system shown in Fig. 2 is developed to measure and analyze the throughput of wireless network, according to measured distance while moving one mobile node in this network.

Figure 2. System architecture

The vehicle is moved in WiFi network as mobile sensor node. It is constructed by using ultrasonic sensor and ESP8266 NodeMCU WiFi module. Ultrasonic sensor is used to measure the distance when mobile sensor node is moving. Blynk application is implemented to control mobile sensor node and it can also command desired movement of node deployment. NodeMCU from the mobile sensor node, motor controlling via phone, two laptops (one is for sensor data monitoring and the other is for network performance monitoring) are connected to WiFi via fixed router. This router is configured by IEEE 802.11n wireless LAN. The sensor data (distance measured) from the ultrasonic sensor is sent to ThingSpeak sever. NodeMCU transmits these sensor data to ThingSpeak via the router shown in Fig. 3.

And measured distance data is monitored with one laptop that is connected to WiFi via the router. The throughput and signal strength of the network is measured and it is monitored by using WinBox software shown in Fig. 4. From these data, the mobility pattern of this network is analyzed according to measured range information from ultrasonic sensor.

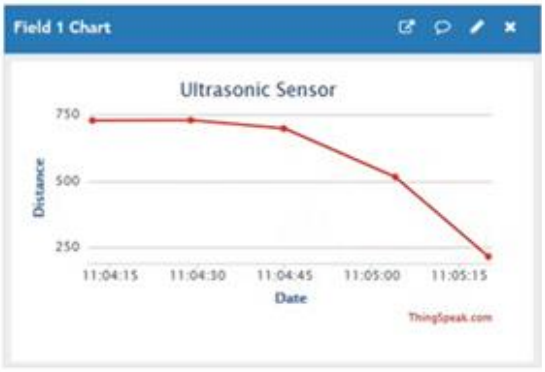

Figure 3. Monitoring moving distance on ThingSpeak

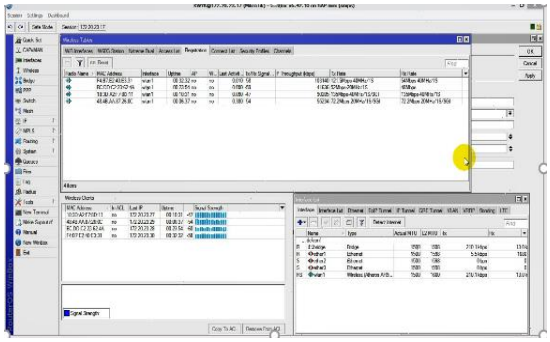

Figure 4. Monitoring throughput and signal strength

B. Testing Environment

In this work, 30 feet path from starting point to end point (at fixed router) distance is specified between fixed router and mobile sensor node. Testing environment 1 is less friction than testing environment 2 shown in Fig. 5. For these two testing, constant speed of vehicle is set as 10 cm/s. Ultrasonic sensor HC-SR04 can be used to measure distance in range of 2 cm-400 cm with accuracy of 3 mm. This sensor module is needed to trigger for signal transmission by using NodeMCU. NodeMCU reads arrival time between triggering and received echo. The speed of sound from ultrasonic sensor is around 340 m/s. When the system starts, sensor releases ultrasonic waves and gather return echo after hitting the obstacle. Based on round trip time, distance can be calculated.

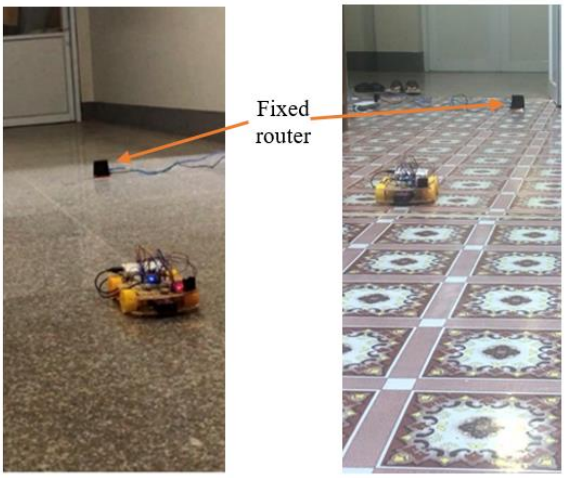

Figure 5. Testing in environment 1 and 2

C. Discussions

By analyzing test results within one-minute data collection, distance 1 from mobility of vehicle on the floor with less friction cannot be the same as actual distance. Distance 2 from mobility of vehicle on the floor with more friction can be approximately the same as actual distance shown in Fig. 6. In Fig. 7 and Fig. 8, signal strengths and throughput from two different testing are plotted with collected data time. Based on these results, changes in signal strengths and per-node throughout is approximately the same for two different testing. It can be seen that changes in signal strength and throughput are approximately constant in random direction mobility model. Reason for testing with constant vehicle speed in this mobility testing is to be focused on friction and frictionless environment. If variable speed is used in testing mobility, it is necessary to consider several parameters for suitable control technique and stability. Thus, this work was initially implemented and tested with mobile vehicle node without variable speed control program.

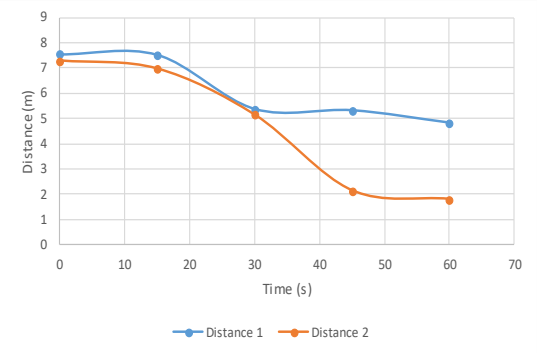

Figure 6. Results of moving distance in environment 1 and 2

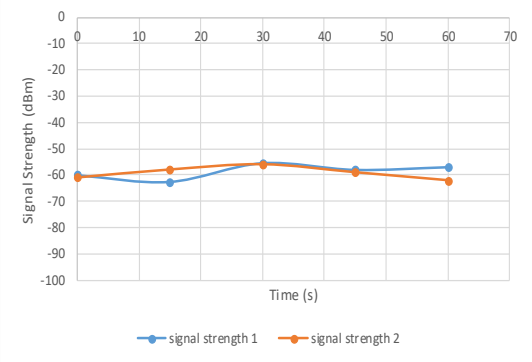

Figure 7. Results of signal strength in environment 1 and 2

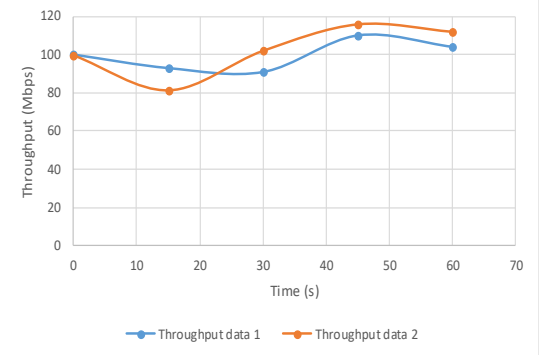

Figure 8. Results of throughput in environment 1 and 2

Based on measured results, per-node throughput changes gradually with distance except at initial start and final stop of mobile vehicle shown in Fig. 9 and Fig.10 in testing of two environments. The throughput at each distance seem nearly the same because vehicle mobility is stable in limited small range. Signal strength is almost stable with time and distance for two environment shown in Fig. 11 and Fig. 12. In this case, vehicle control is needed to consider for stable motion along defined path. It can be seen that network throughput may be affected by measuring moving distance collected from ultrasonic sensor equipped with moving vehicle in WiFi network. The IEEE 802.11n support a maximum theoretical throughput of 600 Mbps. But some routers can support theoretical maximums of 150, 300 or 450 Mbps depending on their configuration. The measured throughput from two different testing is in the range of 80 to 120 Mbps. It is lower than maximum theoretical throughput. This would be occurred when there will be some factors such as interference, packet delay and network workload.

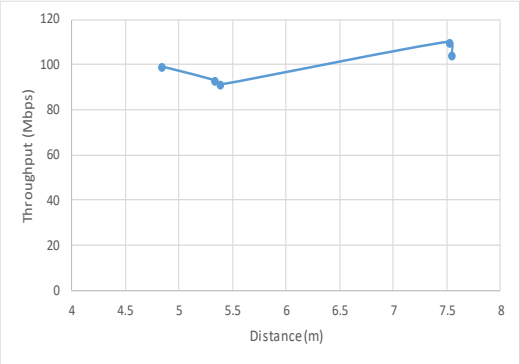

Figure 9. Plotting throughput versus distance for environment 1

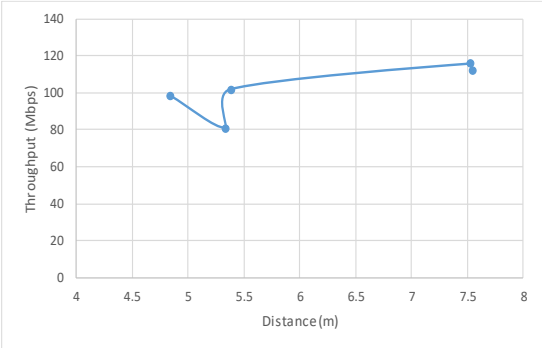

Figure 10. Plotting throughput versus distance for environment 2

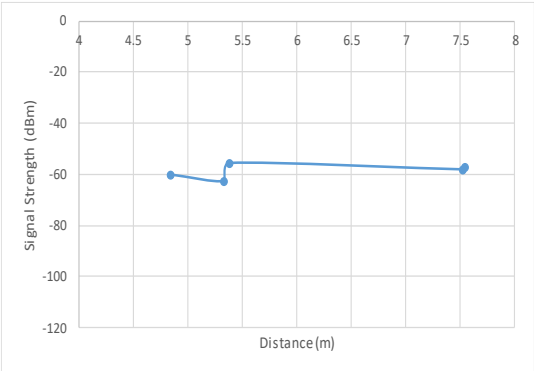

Figure 11. Plotting signal strength versus distance for environment 1

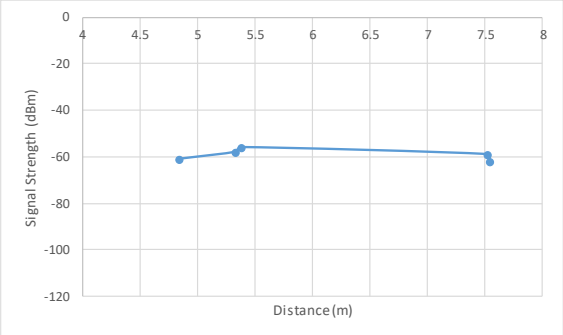

Figure 12. Plotting signal strength versus distance for environment 2

V. CONCLUSIONS

In this paper, range-based localization was used in measuring vehicle mobility in wireless sensor network. Measured range distance is obtained from ultrasonic sensor equipped with vehicle moving in WiFi network. The mobile sensor node was constructed with NodeMCU, WiFi module and ultrasonic sensor as localization finder. In vehicular mobility, all mobile sensor nodes can move arbitrarily fast. To maintain network performance for some constant time, experimental tests are done with constant speed of mobile sensor node. It is still needed to consider modifications for realistic vehicle mobility. Changing behavior of signal strengths and per-node throughput are presented and discussed on two different testing environments. Extended study on wireless mesh network by using many mobile sensor nodes can be applied on massive wireless IoT.

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Information Security Risk Management in Electronic Banking System

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Abstract

Information and communication technology stands up as an important and effective part of various organization in Myanmar. By use of telecommunication network and internet, the information security and IT risk management system become indispensable requirement of organization. The organization need to maintain their information from when internal or external threats are incident. All of threats might control the information security risks and organizations are managed their own information environment. So, the organization required that they maintain and cover with high level security system. This paper focuses on information security risk management in the some activity of electronic banking system (e-banking system). The electronic banking technology is allowed for a variety of implementing financial services between organizations and customers. It allows serving, business market to improve customer service all the time [3]. The use of e-banking system is high rate in developed countries and comparatively lower in less developed countries. The analysis of the evolution and existing standard of electronic banking made the economic opportunity grow for the government [4].

Keywords: *E-banking services, security, risk management*

I. INTRODUCTION

All organizations must have information security model and it is a resource or information to be protected and kept safely. Assets, vulnerabilities, threats, and controls are the basic requirements of information security model. E-banking (Electronic banking) technology is popular in Myanmar banks. It a means of communication between computer system and information communication technology to archive many benefit for customer and bank. There are a total of twenty eight local banks and thirteen foreign branched banks in Myanmar. Including five organizations authorized by the Central Bank of Myanmar to provide mobile financial services. The way for customers to interest with banks, E-banking channel, such as online banking channel, mobile banking channel self-service terminals, social media platforms and mobile payments channels are very interesting. Many banks in Myanmar have long offered e-banking services for customers to perform online balance enquiries and fund transfers need to solve the requirements to assess customers' compliance [3].

By the increasing use of mobile devices and social media platforms, banks depend on the requirements of technical and controls corresponding e-banking service system platforms have been enhancing their existing e-banking system platforms to improving their mobile banking application system and functional platforms. Professional responsibilities of information security professionals include a mix of technical and nontechnical activities. Their technical responsibilities include planning, implementing, upgrading, or monitoring security measures for the protection of computer networks and information. It is important to identify asset that are need to be upgrade process in developing technology and potential risks with e-banking system and so that the organization can deploy the right controls across the banks. The challenge is to determine what needs to be protected. By means of assets manages by IT systems, which have one or more vulnerabilities. Adversaries are interested in exploiting these vulnerabilities by means of threats. Information security professionals use control to ward off these threats.

In developing mobile technology and the potential risks associated in present, the system need to be upgrade process a wider scope of e-banking process. The new requirements shield of online banking, mobile banking, self-service terminals, e-banking services in social media platforms and contactless mobile payments detail depend on the technical requirements and controls corresponding to the provision of e-banking services. Much of new risk management are required to solve and guide customer complain. The weakness of the requirements for

banks has an extended scope of e-banking. Banks will need to perform an overall review based on the new to ensure compliance with the extended scope of requirements. In the technology control standard of the industry improving technology risk and actively issued adapt increase regulate on cyber risk and technological risk. Banks are used this process to assess the benefits carry by e-banking channels and improve upcoming technology strategy and way [3].

The other researchers work related in e-banking system are top point out the risks of e-banking which both banks and their clients face all the while placing special attention on examples of risk management of electronic banking and security challenges in e-banking. The objectives of this survey analysis are to understand the impact of e-banking on the banking performance, to know the various risks and security challenges in e-banking, to manage the risk and security aspect of various e-banking services where customers have high level of concern and to get knowledge of the e-banking and its impact on traditional services in less developed countries.

II. E-BAKING AND SERVICES

E-banking is providing to twenty four hours access available at every time for whom with the customer contact. It can reduce waste time to visit to check information of balances or transferring money to another as in normal bank. Because of low operating costs of e-banking came into existence in greater numbers of the usage of e-banking by the enterprises now. E-banking transaction service is based on information technology and it can provide financial transactions electronically fund process faster services and low cost between banks and customers. Automatic teller machines are start of developing banking process and it developed to internet banking services done in mobile devices become used of the best financial transactions. It can use easier and banking process faster between customers and banks. To perform financial transactions by use of card, the electronic payments and where the cardholder pays by using of computer systems. There are many transaction types such as deposit, withdrawal and electronic fund transfer to account. To protect non-financial transactions, the administrative that including identification number is needed. During e-banking procedures the electronic funds transaction need to be activated. TV banking, short message service banking, mobile banking and internet banking system are useful types of e-banking system. For customer to access financial service an internet connection is required. E-banking remove the customer personnel activity in e-banking system, the service transaction to provide the benefit depend on customer responsibilities. Thus, to fit the process the customer require knowledge, understanding the technology of the banking system and interface.

In development of mobile technology, mobile software is a program that downloaded onto a mobile device or accessed by a device used of the internet. The use of mobile device for financial transaction service need to understand the sequence of instruction. The knowledge of the risks and experience of customer are limited. The mobile banking applications (online shopping systems, mobile payment systems, mobile banking system, mobile play store and so on) are applications that can be used by mobile devices that allow to browsed complete customer wish and banking transactions process. Mobile accounting, mobile service fee and user financial information are the three main parts of mobile banking system. Administer managed to operate of the account in mobile accounting services. Mobile accounting services have account operations and account administration. Account operations are services for fund transfers and bill payments. Account administration manages by blocking lost cards, update active accounts, and instruction to check. Mobile service fee are the services that are required to benefit for an investment account, including the variety of funds operation and requirements of information securities. Account financial information and market target area are the main part of mobile user financial information services. Account information includes information of balance inquires, requests, alerts, location of branch, and veritable card information. Target area information provides information played exchange rates, interest rates, products and services information. Many banks in Myanmar have mobile application that allows to taking online banking application in mobile devices. It more convenient and quickly checking up to customer account information and funds transaction.

Every bank has system architecture for managing the operation and security risks depend on system design and control processes. Bank also need to be update and the staff require training to new system architecture for bank efficient service. The important critical issue for banking system is reputation. The common type of risk in e-banking is transaction risk. Because of incorrect processing, data integrity compromising and by access of unauthorized

to system the transaction risk can occur. The missing authorization vulnerability happens when a software program allows users access to privileged parts of the program without verifying the credentials of the user. This vulnerability is particularly harmful in the financial industry. Attackers are always trying to find parts of financial information systems that they can reach without credentials. For example, according to the "top 25 dangerous errors" publication, hundreds of thousands of bank accounts were compromised in May 2011 at Citigroup as a result of missing authorization vulnerability [1].

The process of risk management and implementation of business objectives are important in information system.

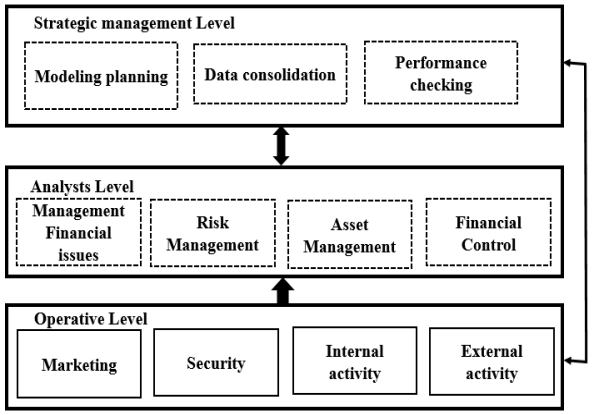

Figure 1. Architecture of Banks' information management system

This system structure is included three levels of organization, which are strategic management level, analysts' level and operation level of organization which can be seen in figure 1. The way of efficient transferring information between management level and strategic management level can be easy to achieve goals and operation process including the expert's field of risk management, analyst state to offer solutions in banking operation. The middle of management specific set of goals and recreate the specific issues.

III. SECURITY OF MOBIL-BANKING

Today mobile communication becomes network connector of the human society. Mobile banking system is one of the most useful applications in mobile communication system. The information security is an important part in mobile banking system between customer and banks. The information security in mobile banking system is be as follows:

A. Information distortion and damage

Mobile communication system is used wireless transmission media the modulation process need to convert analog data to digital signal modulation techniques. In mobile communication network technology provides limited tools to cover the transmission media. Attacker can modify the information to distortion or damage the used of legitimate users the overlapping and installation acceptable in mobile communication network and then modify or update data or delete. Confidential banking information may be weak and distortion damage in the daily transaction devices [4].

B. Incomplete process information

If a customer use mobile device in poor of communication network connection it will be become unwanted data processing and incomplete communication information. Delay or failure transaction can occur in interruption by noise or other signals. It lead to unfinished transaction process could be easy to incomplete data or loss. Banking organization need to turn off and make the information unfinished.

C. Virus attacks control

Target of hackers is the weak of e-banking and security is one of the risk management problem by customer in access their account using internet. An intended as an attack have plan to the system fraud. Living in digital age, despite the current virus on mobile operations found mainly destruct mobile phone function, use of mobile phone application, restore the battery of charges, log on to using internet and record other data are all faced with the threat of exposing private information [2]. The virus designed to do harm to mobile system, replicated and infected to operating system and spread to other system. By the passwords control, firewalls and antivirus software control are built into the information system itself by technical controls [1].

D. System and data integrity

E-banking systems are more available than Traditional banking system. It can provide internet access for their customers with paper less system. A customer can print the information by using internet. It provides the relations and satisfaction able to create. If a customer does not want to use e-banking system, the system may be break and cut available services to

a customer wished in bank. E-banking system allowed to customer the control and manages in his account. The integrity process of the system allowed testing and documentation to complete of banking system.

E. Digital Signature

By means of digital signatures based on encryption and decryption method technology which built solution with signed document verified the authenticity of signed record. It plays a role in the data authentication and non- repudiation.

IV. Cyber Security Risks

Cyber security risks in e-banking systems are corresponding to using of internet. Cyber security in an organization is used of technical infrastructure to protect data and information from cyber-attacks on e-banking system. Source of cyber risk can meet, the organization reviews to manage the weakness of issues and IT function cover to multi-level of cyber risks.

The following resources are the key areas that organizations should look into when incident of cyber risks.

1. Leadership and governance
2. People factors
3. Information risk governance
4. Business management
5. Operations and technology and
6. Rule and law

A. Leadership and Governance

Maintenance and development of the e-banking system depend on the technology and responsible in department of organization. E-banking system with build in designed, infrastructure of process and procedures. That have to be responsibility in e-banking system to obtain manage the procedures of reasons in nonfunctional conformance policies. The departments need to develop and manage the risk of requirement. If necessary to assistance formulation and procedures should checks the adequacy existing controls management.

B. People Factors

All of employees in an organization have the various knowledge and expert in their job. The responsibilities of senior management in e-banking are abide in security and banking strategy. By using IT to handle not efficient manage to organization them self. To improve the weakest of cyber security in people, training the staff to improve knowledge and understanding help are the effective ways.

C. Information Risk Governance

The requirements of evolution continuous corresponding to the risk management in organization depend sample review the procedures and policies of system. That identified requirements of risk assessments on e-banking system sample document and result in the risk. Documentation procedures and policies performed required activities. In use of different e-banking frame, the requirements and industry achieved the good banking policies in time by time.

D. Bank Management

The process of managing the bank's activity refers to bank management. It is financial relations connect with bank activity and management function in implementation of banking system. The application infrastructure support to target areas that are with a view to get profits and recovery plans should failure of other technical issues. The operation management for organization is to manage daily improves all over the customer.

E. Operations and Technology

The development of internet technologies with new processing of frame including mobile banking, online banking and media form submit to web site. The risks occurrence in banks, it is associated risk assessment to step by step to design banking frame. The weakness of cross channel risk assessments is the one of the common factor in regarding e-banking system. The impact arising from other banking system for bank to comprehensive is important for cross channel to understand on banking system. Mobile application updates are software updates that fix activity with components of the application software. The system directly developed and released by software and it automatically checks for installing or updates system administrator intervention.

F. Rule and law

E-banking system determine to perform normal risk assessments on existing banking system is required. Compliance departments work in financial services activity meet for efficient, transparent and markets are fair. The documentation is reducing system risk and financial crime. The

independent assessment is needed to decide the various technology requirements in internal or external consultants to generally review their requirement and compliance.

V. METHODS

A policy is a document that records a high-level principle or course of action that has been decided on. An information security policy therefore records high-level principles on information security that have been agreed on at the highest levels of the organization. The goal of an information security policy is to obtain endorsement at the highest levels of the organization for information security activities. Policies are written in a language that is general enough to deal with routine developments in business and technology. While a policy specifies a general direction for the organization to follow, without concerns for how to get there, standards, guidelines, and procedures focus on how to get where the policy desires to go [1].

Risk is a quantitative measure of the potential damage caused by a specified threat. We may write this managerial concern as:

Manager's decision problem = max (profit) or
Manager's decision problem= max (revenues – cost)

At the very high level, risk management can be known as the management the financial impacts of unusual events. To modify the manager's decision problem as:

Max (Revenues – cost – Δ),
Where Δ is the impact of unusual events on the organization.

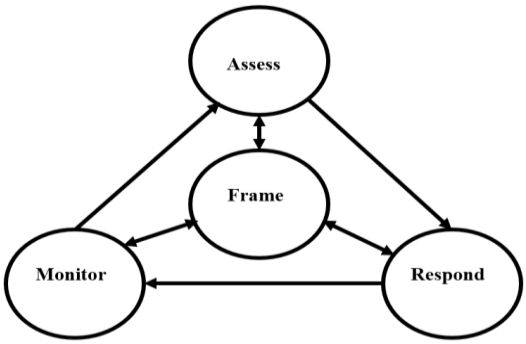

Figure 2. Risk management framework

The risk frame, the risk assessment, the risk response once the risks are assessed and ongoing risk monitoring based on the experiences are the four components of risk management framework show in figure 2. The frame determines the risks the organization focuses on. The assess stage quantifies the risks in the risk frame. Monitor and respond involve managing the assessed risks. A framework is a structure for supporting something else. In the management literature, frameworks are used when a large number of ideas are to be organized in a manner that can be understood and memorized by many people. The objective of framework is a recommendation for managing information security risks. The identified organizational risks can include many types of risk. The risk frame establishes the context for risk managements by describing the environment in which risk based decisions are made. IT risk management is the assessment, monitoring and response to risks associated with the use of information systems in an organization [1].

The IT risk frame establishes the context for risk management by describing the environment in which risk-based decisions are made. The frame clarifies to all members in the organization the various risk criteria used in the organization. These criteria include assumptions about the risks that are important, responses that are considered practical, levels of risk considered acceptable and priorities and trade-offs when responding to risks. Risk framing also identifies any risks that are to be managed by senior leaders/executives [1].

During in (2018 June to December) the past six months, efforts by KBZ Bank staff across the country bring one millions of people to the mobile-friendly economy. KBZ Bank aims to reach 30 million KBZPay users over the next ten years, as mobile population growth in the country increases [7]. Because of the average of over two thousands KBZPay customers increase in monthly by Kanbawza bank (3), this research is used of case study based on analysis of information risk management in banking. The optionality question are designed to analyze the risk associated with in the banking sector. The user option are paper form and administered to local zone in University of Computer Studies (Myeik) in Myanmar. The total number of 135 students including employee respondents was achieved.

The five scale rating structure system includes rarely option, never option, very frequently option, frequently option and occasionally option. In order to obtain the risk impact associated to options as contained in the administered optionality. The marge of rarely and never option values indicate the high risk impact, the associated option values show medium risk impact and the marge of very frequently and frequently option vales indicate low risk occur [8].

A. Results and Discussion

This research state that 28.15% are male and 71.85% are female respondents. It indicate the most of respondents banking user of female are more than male banking user because of female students and employee more than male show in figure 3 below.

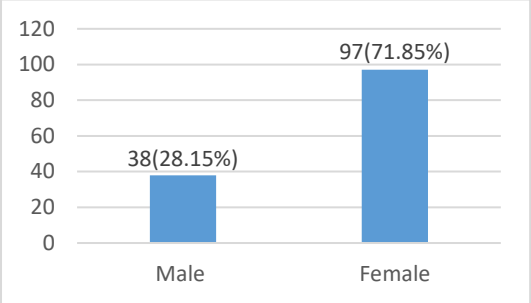

Figure 3. Gender distribution

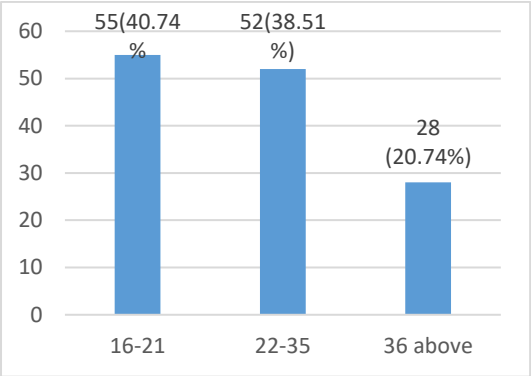

Figure 4. Age distribution

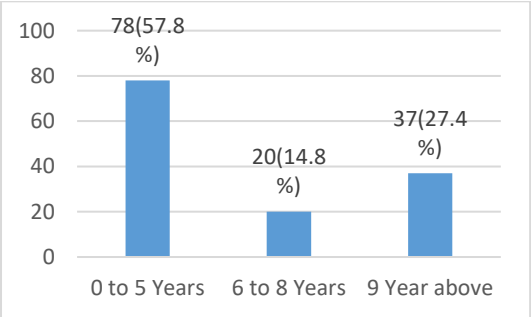

Figure 5. Banking experience

Figure 5 state the experience in banking system, 57.8% have under 5 year experience, 14.8% respondents have 6 to 8 year experience and 27.4% respondents are with 9 year above. This indicate all respondents with a good period of time in banking system.

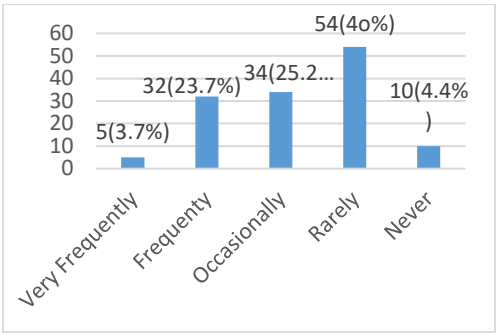

Figure 6. Confidentiality level frequency of KBZ pay and ATM

In above figure 6 states the ATM (Automated Teller Machine) card and KBZ pay account exposed 40% of the rarely live their account, the respondents occasionally leave 25.2%. The frequently never used with 4.4% and 23.7% respondents never leave their account and card. 3.7% of the banking user leave their account very frequently. 21.48% respondents are occasionally.

The figure 6 shows the state of seek for assistance during an online transaction. 62.9% of respondents no need to seek for online transaction, 17.1% occasionally seek for assistance and 20% of respondents seek for assistance.

In figure 7 in below showed 31.85% respondents used their devices in financial process frequently and 29.63 very frequently. 21.48% respondents are occasionally and 17.04% respondents are rarely used their device in financial transaction.

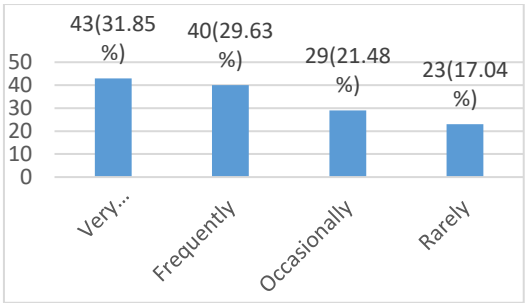

Figure 7. Frequency used of smart devices with password in financial transaction

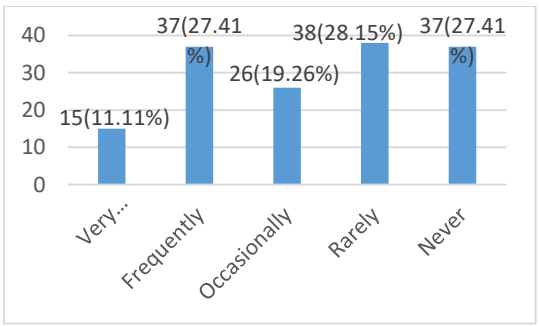

Figure 8. Frequency of free wireless access point used for online transaction

In figure 8, it shows the use of wireless access during in online transaction. 28.15% are rarely use wireless access point, 19.26% used occasionally, 27.41% respondents are never use free wireless access point in online transaction, 27.41% are frequently used and 11.11%respondent very frequently use wireless access point for online transaction.

According to figure 6,7, and 8 in this survey, that exposing ATM Card or KBZPay will have a high impact of risk on the customer, while sharing of ATM Card and PIN with third party also has a high impact risk on the customer, while the necessary of devices using password for online transaction and having a licensed antivirus software with on the devices used for online transaction indicate a low impact risk, during online transaction, using a free wireless access point and been debited without a successful transaction all have a high impact risk on financial institution.

B. Suggestions from Survey analysis

1. In e-banking system construct by one of supervisory policy manual and other monetary authority hold together with two factor authentication services, account unity response services from when the threat is incident and so on. The customer compliance of requirement is solved by banks. Banks need to examine to the e-banking services in supervisory policy manual.
2. The board of organization and senior managed to e-banking system when the incident of risk occurrence and including employee resource. The requirements of adequate employee skills are depend on expertise senior management to manage the risk. Accounting staff manage e-banking services for specify of banks.
3. Fund transfers to unregistered third parties are considered high-risk transactions and should be subject to two-factor authentication. Small value fund transfers to unregistered third parties can be performed without two-factor authentication. Banks should refer to the requirement stated in the additional guidance on SPM (Supervisory policy Manual) also consider the banks' own risk appetite when determining the caps for small value fund transfers [3]. In two factor authentication, the fund transfers to without accounting person to performed risk transaction. It can be performed by banks should supervisory policy manual to manage the risk, bank also determining to allowed for few value of fund transfers.
4. Application forms are not required in online. Banks are needed to solve to assess the risk and establish controls the online services information. The required of service include:
    a. The use of password to protect the confidentiality and integrity of the information sub-post to online.
    b. Manage to cyber attacks through the documents information.
    c. If required, checks to rule identity of the customer online services information [3].
5. In mobile banking system, the security and weakness of mobile application specific risks and banks also need to assess the detail risk and management to mobile banking infrastructure. Many risks can be asses in attacker attack, risk from virus, structure of security and devices loss of customer.
6. Including the weakness of credit card information and transactions process run in mobile devices, banks need to assess security risks on contactless payments and defined multilevel security risk. The standards of security issued depend on banking associations and mobile payment services are ensured in banks sector.
7. The notifications system is important part of the maintaining and solving the process of system implementation. Banks always need to promptly customer requirement notify transactions include card damaged or other. Banks need to solve immediately by using automated process after detection notify high risk transactions occurs.

VI. Conclusion

Many organizations use information and communication technology. For developments of any organization (such as business, financial, database record keeping and so on) are needed to manage information security and risk management system in organization. Especially, many banks in Myanmar use information communication technology. Every banking system to fulfill the needs of the customers in managing their personal information, data, and security. The e-banking technology is allowed for a variety of implementing financial services between organization and customers. Because of mobile banking system is attractive and convenient to

perform remote banking approached, the improving information security risk management in e-banking system, it can useful tool of system development and main issues of mobile telecommunications technology especially mobile device function improved [4]. It can integrate the mobile banking and current service. In banking industry, it make easy use of the benefits provided by of mobile phones and develop a unique customer oriented services will be stand to play more standard role. The main challenges in e-banking system are it has run and developed for different operating systems of mobile device and on security issues which has the risk to the customer who use mobile banking system.

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Data Mining

DATA MINING TO SOLVE OIL WELL PROBLEMS

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Abstract

The purpose of this work is to create a learning algorithm which is based on accumulated historical data on previously drilled wells. Wells will forecast an emergency accompanied by drilling. Such a decision support system will help the engineer time to intervene in the drilling process and prevent high drilling costs simple and repair equipment resulting in an accident. The article provides a brief overview of the most common method of artificial intelligence — artificial neural networks, as well as the main areas of their application in the oil and gas sector. In their work, the authors distinguish three main areas of use of such technologies: interpretation of geological data, exploitation of deposits (smart fields) and price forecasting. The use of methods based on artificial intelligence increases the efficiency of the work carried out both in exploration and production, makes it possible to achieve better results with less cost.

Key words*: classification oil and gas, drilling complications, machine learning, neural network, effciency improvement, gradient boosting.*

I. INTRODUCTION

In the course of the study, work was carried out on the study of machine learning methods; review of existing practices for the use of melon models to improve drilling efficiency. The provided reports on drilling of wells at the field are analysed. Identified wells in which there were complications. Calculations on various machine learning algorithms are carried out to identify the algorithm that gives the minimum percentage of error. As a result of the study, a model based on gradient boosting was calculated to classify complications in the drilling process.

II. ARTIFICIAL INTELLIGENCE IN DRILLING

The requirements of the practice of drilling deep oil and gas wells require a broad range of needs for the theory. In this case, the theory should explain the flow of drilling technological processes both in typical regimes and at the time of the onset of complications and during development, treating complications as an integral part of such procedures. It is desirable that a theoretical description of difficulties allows judging them not only at a qualitative level, but also quantify the interrelation of their essential variables. Because of the rather narrow applied nature of modelling tasks and prevention of complications, their formulation and evaluation of the results obtained should first of all be guided by the needs and possibilities of practice. The existing capabilities of computer technology make it possible to carry out calculations that several years ago seemed laborious. Let us briefly review the existing works, which were aimed at improving the drilling process using neural networks and machine learning.

At present, methods of neural programming networks for solving problems in various fields have been widely used. An artificial neural network is an interconnected group of nodes, similar to our brain system. Figure 1 shows the neural network scheme. There are three layers, each circle is a neuron, and the line is a connection between neurons. The first layer has input neurons that send data through communication lines to the second layer of neurons, and then through a large number of link nodes to the third layer of output neurons.

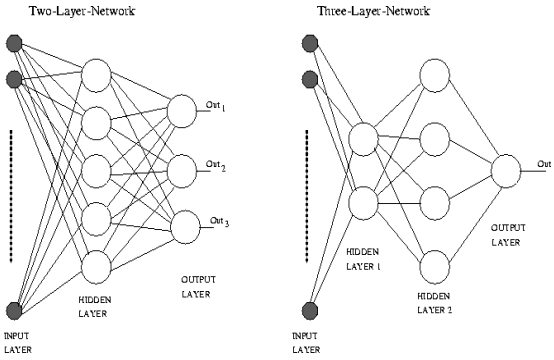

Figure 1. Diagram of the neural network

III. THE ALGORITHM WITH HIGH ACCURACY OF THE CLASSIFICATION OF PROBLEM

According to the algorithms of machine training given in the previous chapter, calculations were made to classify (forecast) the complications in the drilling process. For calculations, the Python programming language was used. Percentage between training and test sample 65/35%. The training sample is a sample based on which the chosen algorithm adjusts the dependency model. The test sample is the sample by which the accuracy of the model used is checked. Metrics were used to assess the quality of the models used to classify the complications in the drilling process. For each of the algorithms, precision (precision), recall (completeness), and F-measure metrics were introduced.

$$\text{precision} = \frac{TP}{\text{TP} + \text{FP}}$$

$$recall = \frac{TP}{\text{TP} + \text{FN}}$$

where,

TP - positive observation, and expected to be positive;

FN - observation is positive, but it is predicted negatively;

FP - observation is negative, but predicted positively.

Precision is a kind of share of objects, which is called a positive classifier, and in this case, these objects are in fact positive. The recall is a metric indicating which fraction of objects of a positive class from all objects of a positive class found an algorithm (Figure 2). In other words, precision does not allow you to assign all objects to one class because, in this situation, the FP level will increase. Recall shows the possibility of the model to define this type in principle, and precision - to distinguish the class from other classes.

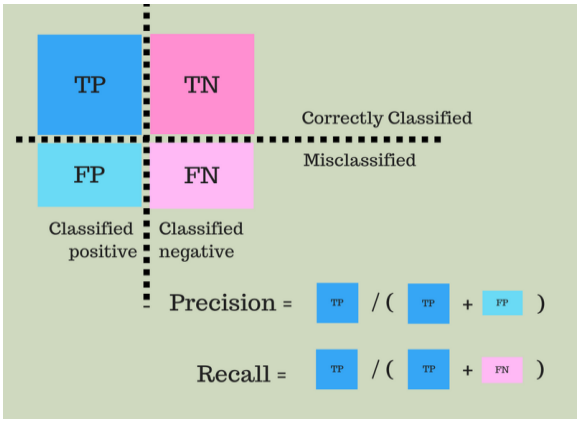

Figure 2. Accuracy of the classification of problem

The F-measure is an aggregated quality criterion that combines precision and recall-average harmonic precision and recall.

$$\text{F} = (1 + \beta 2) \cdot \frac{precision \cdot recall}{(\beta 2 \cdot precision) + recall}$$

where, β is the weight of the accuracy of the metric (β = 1 is the average harmonic). F-measure for completeness and accuracy reaches a maximum of one, and if one of the arguments is close to zero, it tends to zero. The data loaded into the Python software environment was loaded with a single dataset, with a preliminary classification of the complications, for the subsequent learning of the model. As indicated, the training sample is 65% of the total data, and 35% of the data is used to verify the correctness of the model set. The following drilling parameters were used as input parameters:

- Level of tank 02;
- Input flow;
- Weight on the hook;
- Torque on the rotor;
- Rate of penetration;
- Volume of tank 02;
- Gas content.

As a result of the calculations, the following metrics were obtained, for the subsequent detection of the most accurate model.

IV. SUPPORT VECTOR MACHINE

The General idea of an SVM is to solve the problem of correctly classifying a set of data and maximizing a geometric field. There can be multiple separating hyper planes, but there is only one separat-

ing hyper plane with maximum geometric indentation. A direct explanation for maximizing the geometric field is that the hyper plane with the maximum geometric indentation derived from the classification is equal to classifying the training data by a sufficient certainty factor [7]. It is necessary not only to classify correctly, but also to separate the nearest points with a sufficient coefficient of reliability. This process can provide certain data with a good predictive ability called generalization ability.

When solving a nonlinear problem after converting to multidimensional space, it is usually difficult to find a hyper plane that can completely separate the data points, which means that there are some special points. But after removing these special points, most of the points become linearly separable. To solve this problem, we import the sliding variable into the training sample. In a soft-edged situation, the SVM learning task will look like:

$$min_{w,b,\varepsilon}\frac{1}{2}||w||^2 + C\sum_{i=1}^{N}\varepsilon_i. \quad (1)$$

$$\text{s.t.} \quad y_i(wx_i + b) \geq 1 - \varepsilon_i \quad (2)$$

where C is the penalty parameter. Increasing C also increases the penalty for classification errors. You must adjust the target function to minimize the number of singular points while maximizing the offset from the hyper plane.

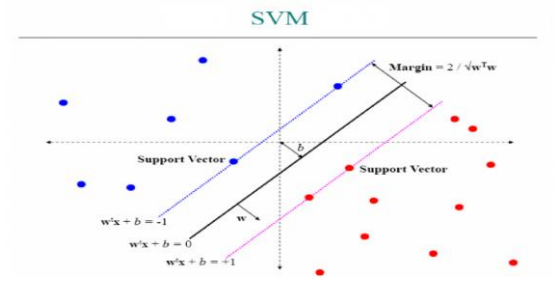

Figure 3. Support Vector Machine

V. LINEAR LOGISTIC REGRESSION ALGORITHM

The linear logistic regression algorithm is a classical classification method in the study of statistics related to the linear logarithmic model [8-9]. This classification model is a conditional probability distribution P (Y / X), which is a judgment model. It can be obtained from the linear regression model hw (x)= $W^T X$ and the sigmoid curve:

$$P(Y = 1|X) = \frac{1}{1+e^{-wx}} \quad (3)$$

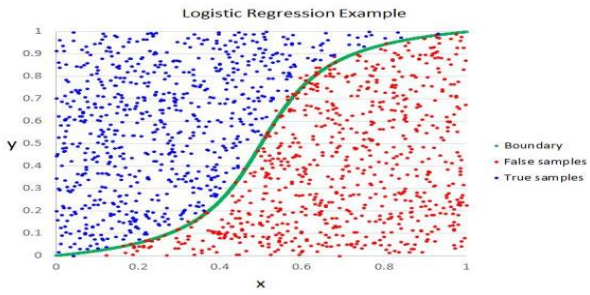

Figure 4. Logistic regression distribution function and density function

$$P(Y = 1x) = f(x), P(Y\,0 = x)1 = f(x) \quad (4)$$

Likelihood function

$$\prod_{I=1}^{N}[f(x_i)]^{y_i}[1 - f(x)]^{1-y_i}. \quad (5)$$

Logarithm likelihood function

$$L(w) = \sum_{i=1}^{N}[y_i\, log\, f(x_i) + (1 - y_i)log(1 - f(x_i))] \quad (6)$$

VI. IMPLEMENTATION AND RESULTS OF THE EXPERIMENT

Table 1. Metrics by model

Algorithm	Metrics		
	Precision	Recall	F-мера
Logistic regression	0.00	0.00	0.00
Naive Bayesian Classifier	0.03	1.00	0.06
Method of k-nearest neighbors	0.83	0.64	0.73
Decision tree	0.97	0.87	0.92
Support vector method	0.00	0.00	0.00
«Random Forest»	1.00	0.87	0.93
Gradient boosting	1.00	0.93	0.97
Neural network	0.92	0.51	0.66

Table 1 shows that the following algorithms of machine learning have the highest accuracy: decision tree; "Casual forest"; Gradient boosting.

Next, we considered the number of correct and correct assumptions in the calculation of algorithms. Table 2 presents the case for situations where there are no complications, and in Table 3 the classification of complications is correct. True is the number of correctly predicted values; false is the number of

misplaced predictions. From the data presented, it can be seen that the greatest number of correct and accurate classifications of situations is obtained using the machine learning method gradient boosting. Gradient boosting (Appendix B) allowed with a minimum of the error to classify the complication from the available data set.

Table 2. Accuracy of prediction of a normal situation

Algorithm	Situation	True	False
Logistic regression	Normal	3916	1
Naive Bayesian Classifier	Normal	2484	1433
Method of k-nearest neighbors	Normal	3911	6
Decision tree	Normal	3916	1
Support vector method	Normal	3917	0
«Random Forest»	Normal	3917	0
Gradient boosting	Normal	3917	0
Neural network	Normal	3915	2

Then, input parameters were analyzed by significance category, weighting criteria for gradient boosting. The greatest influence on the operation of the algorithm is "Input pressure", "Torque", "Flow rate at the input".

- Inlet pressure (0.3115)
- Torque on the rotor (0.2709)
- Inlet flow rate (0.2363)
- The volume of tank 02 (0.0704)
- Gas content (0.0601)
- Weight on the hook (0.0160)
- Rate of penetration (0.0082)
- Level of tank 02 (0.0000)

VII. Example of Gradient Boosting Implementation

```
In [16]: # Gradient Boosting
         from sklearn import datasets
         from sklearn import metrics
         from sklearn.ensemble import GradientBoostingClassifier
         # fit a Gradient Boosting model to the data
         clf = GradientBoostingClassifier()
         clf.fit(X_train, y_train)
         print(clf)
         # make predictions
         expected = y_test
         predicted = clf.predict(X_test)
         # summarize the fit of the model
         print(metrics.classification_report(expected, predicted))
         print(metrics.confusion_matrix(expected, predicted))

         importances = clf.feature_importances_
         indices = np.argsort(importances)[::-1]

         print("Feature importances:")
         for f, idx in enumerate(indices):
             print("{:2d}. feature '{:2d}' ({:.4f})".format(f + 1, indices[idx], importances[idx]))
```

Figure 5. Example of gradient boosting implementation

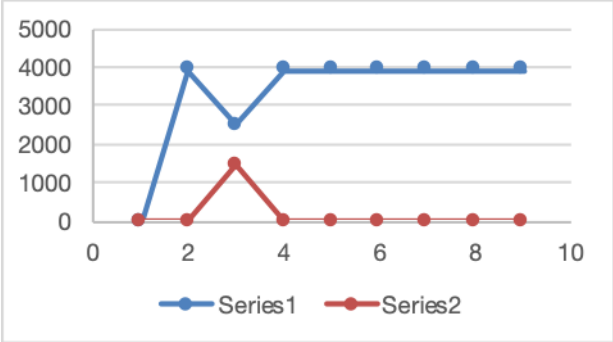

Figure 6. Accuracy of prediction of a normal situation

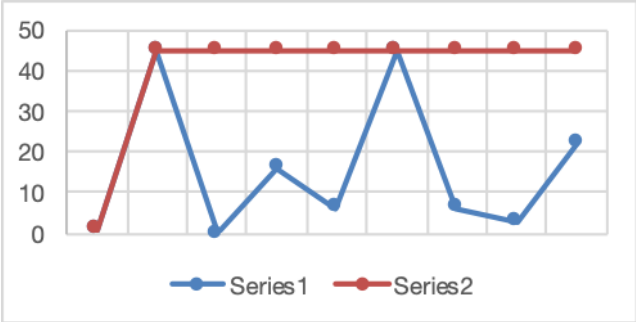

Figure 7. Precision of prediction of problems

VIII. Conclusion

The most effective methods of machine learning were used to classify complications in the drilling process. For the algorithms, drilling parameters were selected, which will be obtained at the entrance to the program, for its training and further prediction of complications. As a result of the calculations, the most effective method of machine learning-gradient boosting-was chosen. This method showed the smallest error in the complication classification test. Of all the drilling parameters, the inlet pressure exerts the greatest influence on the classifier.

As a result of this work, it can be argued that the algorithm was chosen, which with a minimum error is able to classify complications in the drilling process on the basis of the parameters recorded on the rig. Such a program will help the engineer to intervene in the drilling process in time and prevent high costs of downtime and equipment repair. It is recommended for further work on the project to include more wells with complications to more accurately adjust the program for the classification of various complications. It is very important for the oil field technological service to determine the multiphase (oil-water-gas) stream flow regime because, for example, even high liquid flow rate in the not corre-

sponding regime can lead to feed failure of the pump and to emergency stop of oil well. However, if this situation could be classified at the early stage, it can provide to avoid accident situation and thus to maintain oil well operating efficiency.

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Defining News Authenticity on Social Media Using Machine Learning Approach

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Abstract

Social network and online news media are becoming popular in today's era. Due to low cost, easy access and rapid diffusion, social media platform becomes a source to distribute false information. Fake news propagation on social media can cause serious negative effects on human society especially in politic, reputation and finance. So, automatic fake news detection plays a vital role to robust news media platform on social network. Defining news authenticity is insufficient based on news content only. It also needs to analyze social features of news. In this paper, we propose an approach to detect fake news on social media that covers both news content and social context. We use synonym-based feature extraction method and three different classifiers based on multidimensional dataset. Experimental result shows the effective as an accuracy way to define news authenticity on online news media.

Keywords: *fake news, news media platform, news content, social context, news authenticity*

I. INTRODUCTION

Nowadays, the more people use online news media, the less they spend with traditional news media such as newspaper, journal and television due to advancement of latest technology. Social media becomes a global information network and it provides an easy, fast dissemination and less expensive way to access news than traditional news media. According to recent survey, an estimated 3.484 billion people in the world were using the social media and it predicts that the number of social media users will increase 9.1 % year on year[1] .In particular, dissemination of fake information on social media has been increasing in today's world. Popular social networking sites such as Facebook, Twitter, YouTube and Google plus are the primary source of diffusion misinformation on online news media[2] .

Fake news refers to news articles which intent to mislead news consumers in order to damage individual or society's reputation, politics and economic events. Social media users are often less likely to critically evaluate the information shared by other people or confirm their existing beliefs. Therefore, digital false information is more spreading about politics, finance and economy in recent years. Sufficed to mention that during the 2016 US presidential election, social bots spread misinformation on Twitter to confuse people opinion. One interested fake news is "Pope Francis shocks world, endorses Donald Trump for president" release statement that was very popular among social media users and it caused unexpected impacts on election[3]. The another conspicuous example of fake news is that Steve Job's company stock share would significantly decline in a short time due to fake news report of CNN's iReport.com in October 2008 [4].

Detecting misinformation on social media is unique challenges. A news piece on online news media has several characteristics to analyze news authenticity. First, fake news is written to mislead viewers and it mimics real news like topic, writing styles and media platform. Thus, it is insufficient based on only news content to detect fake news. Other auxiliary information including social engagement and stance of news or comments are also useful factors for improving automatic fake news detection. Second, exploiting social engagement and news comments are another critical challenge. Today researchers consider to build multidimensional model by combining news content and social context to detect fake news . News on social media is continuously updating which may not be verified with existing knowledge base because it is lack of corroborative evidence. However, auxiliary information in online news is noisy, big, incomplete and unstructured [5]. Therefore, detecting news authenticity on social media needs further investigation such as defining user authenticity, extracting useful post features and social features. The core motivations of this paper are summarized as follows:

- Automatic fake news detection is an emerging problem on social media and it can cause serious impacts on social, political and economic issues.

- Exploring effective features from news content and social context to analyze news authenticity is also a technically challenging task.
- The prior researches about this problem lack labeled benchmark datasets that covers news content, social context. To better guide the future direction of this problem, new multidimensional fake news benchmark datasets are necessary.

In this paper, we discuss News Application Programming Interface (News API), Graph Application Programming Interface (Graph API) to collect news stories from social media, synonym based feature extraction method and three different classifiers to define news authenticity on online news media. The objective of this paper is to describe the framework of fake news detection system by exploring the correlations of content of news, stance of news and social engagement. It also aims to analyze which features of news stories are helpful for fake news detection and to know the ideas and process of automatic fake news detection system.

The remainder of this paper is follows. In section II, we review related work of previous research for fake news detection system. In section III, we briefly discuss our proposed model and fake news detection methods. We describe nature of dataset, evaluation metrics and performance result of the proposed system in section IV and conclude and discuss future directions of this paper in section V.

II. Related Work

The propagation of fake news on social media can cause negative effects on human society. Therefore, automatic fake news detection becomes emerging research topic and there were many prior researches about this problem. We can consider that online news includes news content and social context. The previous research based on these factors to detect fake news on social media.

The authors described fake news detection problem in online news articles using semantic features and three machine learning techniques in [6]. In this paper, it used real-or-fake benchmark dataset which includes 6256 news articles from kaggle.com. They compared the performance of recurrent neural network, naïve bayes and random forest classifiers based on various features extraction methods such as term frequency, term frequency-inverse document frequency, bigram, trigram and quadgram to perform binary classification of online news.

In reference [7], the author discussed automatic fake news detection problem in two aspects: fake news characterization and fake news detection. In fake news characterization phase, they presented the key factors related to fake news on online media including psychological and social theories of fake news, news content features and social context features of fake news, fake news distributors on social media and types of fake news and differences of news patterns between traditional media and social media. In fake news detection phase, they discussed the approaches to detect fake news, available benchmark fake news datasets and challenging issues for fake news detection system on social media.

The author introduced a fake news detection model using hybrid convolutional neural network. It used large dataset "LIAR" which includes over twelve thousands news short statements with meta information from politifact fact checking website in [8]. Then, they evaluated text feature only in this dataset to detect news authenticity by using various classification models and compared the performance with hybrid model which based text and meta data of news.

In reference [9], the authors proposed a model to detect false information on Facebook news pages that covers news content and social context features on three different fake news datasets. In this system, they described content-based methods for news content only and logistic regression and Harmonic Boolean label crowdsourcing methods for social features. These methods are used depending on the social engagement threshold parameter. This paper is also implemented fake news detection model within a Facebook Messenger Chabot to validate the authenticity of real time Facebook's news.

The authors presented WordNet-based similarity measure to calculate the similarity between sentences by using ontology structure in [10]. They discussed about existing sentence level similarity evaluation approaches, WordNet structure, calculating word similarity, computing sentence similarity based on two example sentences and dataset that includes ground truth for sentence similarity calculation.

In reference [11], the authors discussed different approaches for sentiment analysis and compared the performance of domain adaption using supervised, semi-supervised and unsupervised methods. They also explained how to extract words from social media comments and how to work on sentiment dictionary for opinion target.

III. Proposed System and Methodology

The aim of our system is to define political news authenticity on social media. In this section, we describe the overview flow diagram of proposed system in Fig1.In this system, we collect data from Facebook news posts using Graph API. Each news post includes news content, reactions counts and comments. Before we define the authenticity of Facebook news post, we need to calculate content similarity. Therefore, we fetch political news stories from reliable news websites using News API to check similarity of news content.

The system collects news stories from twelve most reliable websites [12] such as BBC News, The Washington Post etc., to compute content similarity with Facebook news. We use synonym-based (WordNet-based) sentence similarity method to get content similarity score in this system.

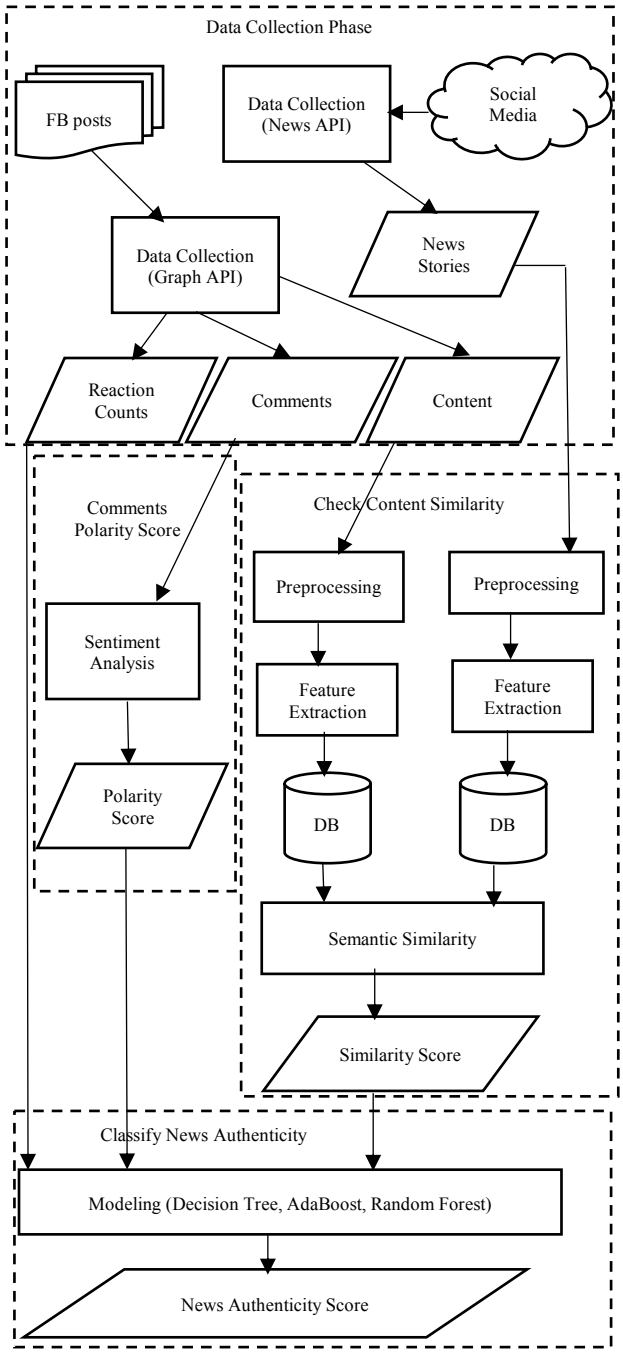

Figure 1. Overview of Proposed System

News comments and reaction counts are also useful factors to classify news as real or not. For instance, when we don't know a news post is true or not by our existing beliefs, we can get clues by reading the news post's comments and post's reactions. To know comments polarity score: positive (news is true), negative (news is false) and neutral, we use VADER: sentiment analysis tool in our system. Then, we define news authenticity by using three different classifiers: Decision Tree, AdaBoost and Random Forest classifiers based on news content similarity score, comment polarity score and reaction counts.

There are four main phases in our proposed system: collecting data, checking semantic similarity between news content, analyzing polarity score from comments and classifying news authenticity. The explanations of each phase are described below.

A. Data Collection

Online news stories can be collected from various social media platforms such as search engine, news websites, Facebook and Twitter. However, it is difficult to know news authenticity manually because there is so much information and it now increased day by day. Thus, building comprehensive large-scale datasets and defining ground truth label of news are challenging task. Generally, determining authenticity of news is based on expert journalists, reliable sources and fact-checking websites. We discuss some available benchmark data repositories below which are focused on fake news detection system.

Vlachos and Riedel firstly proposed a fact-checking dataset for fake news detection system. This data repository includes only 221 news statements. This small dataset includes news content only which does not suitable for machine learning based models [13].

To tackle this problem, William introduced a fake claim benchmark dataset "LIAR" [14]. It is a large scale dataset which comprise 12.8K short statements with annotated ground-truth labels. However, LIAR does not contain the entire news content and it describes only short statement labeled with meta information for each news.

Another dataset called "BuzzFeed" [15] which collected a complete 1627 news articles from nine news agencies over a week in the US election. Every posts and links articles were fact-checked claim-by-claim by five journalists from BuzzFeed news team. The news articles are annotated labels into four categories: mostly true, mostly false, mixture of true and false and no factual content. Nevertheless, it only contained headlines and text with no social features from very few news agencies.

Santa and Williams created a news dataset "BuzzFace" [16] which extends on BuzzFeed dataset in 2018. The news in this dataset is concerned with politic which happened during 2016 US election. It included rich sets of feature such as Facebook-id, post-id, post-url, published-date, reaction count and comments that covered both news content and social context. It also defined news' labels into four categories as BuzzFeed news dataset.

Unlike the traditional news media, online news is real time and it can change at any time. Dynamic social features such as number of reaction count and news comments are become useful features to spot fake news on social media. Before "BuzzFace" is launched, there were no existing available datasets that cover both news content and news social features [17].To tackle the disadvantages of available

existing datasets, we need to build multidimensional dataset which help to develop fake news detection system.

In our proposed system, we use "BuzzFace" news dataset to detect political fake news from Facebook posts. Facebook-id and post-id are useful features which allow crawling news content, comments and reaction of news by using Facebook Graph API. To check content similarity, we use News API to collect political news articles from authentic news sources (websites). By using News API, we get news title, news-url, published-date and content from reliable news websites which are used to check content similarity with Facebook news content.

Graph API: The Graph API is HTTP-based API and it can use to get data into and out of the Facebook platform. It is composed of nodes, edges and fields. We can use nodes to get data about specific object such as Facebook users, photos, Facebook pages or Facebook comments. Edges are used to get connection between objects and fields are used to get data about an object [18].

News API: It is a simple and easy-to-use API for searching and retrieving nearly real time news stories from news sources and blogs all over the world. News content and metadata automatically access via this API using the API key. We fetch news stories from twelve most reliable news sources to check news authenticity. Each news site has own news API to acquire news categories [19].

B. Checking Semantic Similarity between News Article's Contents

News stories from various news websites and Facebook are unstructured, big and noisy. Therefore, these data need to be preprocessing such as tokenization, stop word removal and stemming before checking semantic similarity of news content.

Tokenization: It is the process of splitting a stream of sentence into words phrases, symbols or other meaningful information. Tokenization aims to explore the words in the sentence. The resulting tokens become useful inputs for text mining and further processing.

Stop Word Removal: Stop words are frequently used or insignificant word in a text corpus that will be noise when these words used in text mining process. Thus, the words like pronouns, articles, conjunction and preposition need to remove from the sentence.

Stemming: After tokenization and stop word removal, we need to transform list of token into a standard form. Stemming is the process of transforming a word into its stem word. For example, the words "Speaking", "Speak" and "Speaker" transform into "Speak". Stemming provide classification task more efficient and faster.

After preprocessing steps, we need to calculate similarity between sentences. Similarity between sentences can be measured lexically or semantically. Lexical similarity approach is also known as string based similarity approach which depends on measuring similarity between sequences of characters. Semantic similarity method based on similar meanings of words between sentences. In our system, we implement semantic similarity approach using WordNet to compute similarity score between sentences. This approach is also called synonym-based similarity.

Word Net is a large repository English words which are grouped into synsets (group of synonyms). These synsets are constructed into a hierarchical structure which has conceptual relation between them. The WordNet pointer describes both lexical and semantic relationships between words. To check semantic similarity between two sentences, we first transform the sentences into bag-of-words form T_1, T_2 and create superset $T = T_1 \cup T_2$. After creating superset for two semantic vectors T_1 and T_2, lexical semantic vectors V_1 and V_2 are derived from the superset T. The entry value in V_1 or V_2 describes the semantic similarity of any word in V_1 or V_2 to any word in two sentences. Next, we would calculate semantic similarity score for every word (w_1-----w_n) using the value of corresponding vector entry. Then, small weight value δ ($0 < \delta < 1$) is given to words in the sentence due to their prevalence of usage. Finally, we get news content similarity score by computing cosine similarity from (1).

$$Sim(sentence1, sentence2) = \frac{V_1 V_2}{\|V_1\| \|V_2\|} = \sqrt{\sum_{i=1}^{n} V_{1_i}^2} \sqrt{\sum_{i=1}^{n} V_{2_i}^2} \quad (1)$$

C. Analyzing Polarity Score from News Comments

New consumers' opinion or comments from Facebook news posts help to define news authenticity. Defining comments polarity score is generally based on set of lexicons or pre trained classifiers. In our system, we use sentiment analytical tool: VADER to know users opinion within the comments. VADER is a lexicon and rule-based sentimental tool which intend to extract opinion in social media text [20]. It relies on a dictionary which maps lexical features to know emotion intensities called sentiment polarity score. This score of a text obtain by summing the word intensity in the text. VADER defines how positive or negative score of a text comments in the Facebook news post.

D. Classifying News Authenticity

Fake news detection on social media is a complex and difficult task. To solve this issue, we need to consider efficient supervised machine learning techniques to classify online news' authenticity. In our system, we utilize three different machine learning classifiers namely Decision Tree, AdaBoost and Random Forest classifiers.

Decision Tree: This classifier is a tree-like structure of possible outcomes of a series of related choices. There are three different types of nodes in structure: chance nodes,

decision nodes and end nodes. A chance node is a branch in a tree which shows a set of possible outcomes of the decision tree. A decision node is a decision to be made and an end node is the final outcome of decision tree. The tree starts with a single node which branches into possible outcomes. To select the best attributes in decision tree, it uses two measures: Entropy and Information Gain. Entropy is the useful measure which shows the uncertainly in the data values. It is used to determine how a decision tree can split the data. Information Gain is the measure which is used to choose attribute that best splits the data of each node in a decision tree. The tree is constructed top-down manner by using divide-and-conquer approach.

Although decision tree classifier is easy to interpret, visualize and no need to normalize column, it has few weak points. Decision tree can overfit due to noisy data when a tree is particularly deep. To tackle this problem, we use ensemble machine learning approach in fake news detection system. It is a composite model to get higher accuracy than using a single classifier (decision tree). There are two groups of ensemble methods are parallel ensemble methods and sequential ensemble methods. In our propose system, we use Random Forest classifier for parallel approach and AdaBoost classifier for sequential approach to classify news authenticity. We describe work flow between single classifier and ensemble classifiers in Fig 2.

Random Forest classifier: It is a strong modeling technique which operates a multitude of decision trees and thus improves the accuracy than a single decision tree. Their ability is to limit over fitting as well as error due to bias. It needs adjust parameters such as max-depth, min-samples-split, n-estimators and random-state to achieve the best performance; where max-depth is the maximum depth of a decision tree; in-samples-split is the minimum number of samples to split an internal node and n-estimators is the number of decision trees in the random forest.

AdaBoost classifier: It is used to boost the performance of decision trees on binary classification problems. A single classifier may classify the objects poorly. It is iterative ensemble method which combines low accurate classifiers into a high accurate classifier. The basic concept of AdaBoost is that weak models are added sequentially and trained the data sample in each iteration using the weighted training values.

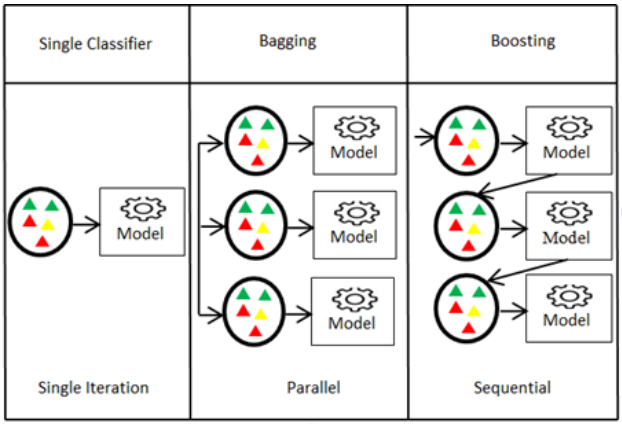

Figure 2. Work flow between single classifier and ensemble classifiers

IV. System Evaluation

In this section, we first describe the nature of "BuzzFace" dataset which are used in our proposed model. Next, evaluation metrics are presented for fake news detection system and we compare the performance result of proposed system using three different classifiers.

A. Dataset

We use BuzzFace news dataset to detect fake news in this system. It contains 1780 news articles with social features which are collected from Facebook news posts. It annotates the labels of the news articles into four categories: 77.81% of the 1780 news articles are labeled as mostly true, 12.87% are labeled as mixture of true and false, 3.99% are labeled as no factual content and the remaining as mostly false. Mostly true and mostly false mean that information in the news article is either accurate or inaccurate. Mixture of true and false is used when the information seems to be accurate or when the information is based on unconfirmed news article by authentic sources. No factual content is chosen when the information is opinion, satire or comics. Each news article includes both news content and news' social features such as reaction and news' comment with annotated label. According to 5-fold cross validation, we split the dataset 80% for training and 20% of the dataset for testing in this system.

B. Evaluation Metric and Performance

The various evaluation metrics have been used in evaluating performance for fake news detection system. In this paper, we apply the following metric to predict the news is reliable or fake.

$$Precision = \frac{|TP|}{|TP|+|FP|}$$

$$Recall = \frac{|TP|}{|TP|+|FN|}$$

$$F1\ score = \frac{2.Precision.Recall}{Precision+Recall}$$

$$Accuracy = \frac{|TP|+|TN|}{|TP|+|TN|+|FP|+|FN|}$$, where

True Positive (TP) means that predicted fake news pieces are actually annotated as fake news.
True Negative (TN) means that predicted true news pieces are actually annotated as true news.
False Negative (FN) means that predicted true news pieces are actually annotated as fake news.
False Positive (FP) means that predicted fake news pieces are actually annotated as true news.

Precision is a good measure when the false positive is high. The news consumers might lose to know news' authenticity if the precision is not high for fake news detection model. Precision shows about how our fake news detection model is out of these predicted positive. Recall calculates how many of the actual positive our model capture true positive. F1 is used to combine precision and recall which can provide overall prediction performance for fake news detection. In our evaluating system, the higher accuracy provides the better performance.

C. Performance Result

This section describes the system evaluation using Decision Tree, AdaBoost and Random Forest classifiers. The experimental results show in Fig 3 by using these classifiers. We observe that the accuracy of Random Forest classifier is better than Decision Tree and AdaBoost classifier. In other word, the accuracy of bagging algorithm is better than single classifier and boosting method in our system.

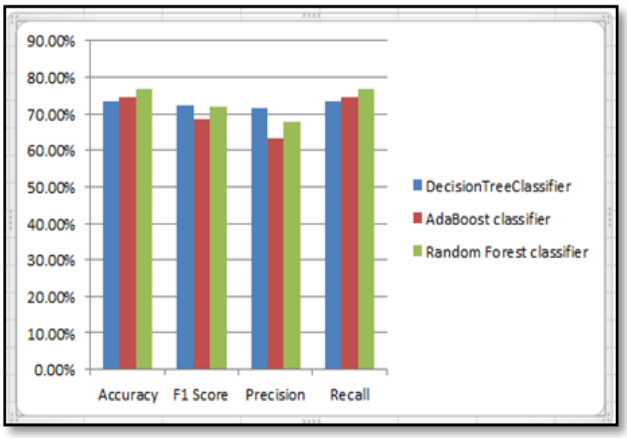

Figure 3. The performance of fake news detection in term of precision, recall and accuracy using Decision Tree, AdaBoost and Random Forest classifiers

V. Conclusion

In this paper, we present a multidimensional fake news data repository that covers news content, social engagement and news stance. We describe how to collect news stories from different news sources using Graph API and News API and how to define annotate label for each news piece. The proposed model in this system is an idea to be more precise and achieve better performance than prior research which based on news content only. Moreover, we perform an exploration study on BuzzFace dataset by applying synonym-based feature extraction method and compare the performance of the system using three different classifiers. In future work, we wish to test hybrid classification methods to get better performance in classifying online news.

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ETL Preprocessing with Multiple Data Sources for Academic Data Analysis

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Abstract

More and more, the needs of academic data analysis are requiring in educational settings for the purpose of improving student learning and institutional effectiveness. In the education world, testing and data are driving decisions for what knowledge and skills students should be learning and how students' learning relates to their learning outcomes. On that occasion, a better option for building a machine learning model is to get effective data preprocessing concepts. For these reasons, this paper describes the very first step of the main research work which considers the correlations between students' academic performance, behavior and personality traits to reveal the presence of an intriguing way. Intuitively, this paper proposes the uses of Extraction-Transformation-Loading (ETL) in the preprocessing stage to collect and analyze of students' data from multiple data sources. In this system, data is collected from multiple data sources based on the structures which are used as a testbed. Students' demographic data and assessment results from Student Information System (SIS), logs of their interaction with Moodle are used for data collection. Then aggregating with Web logs also captures student behavior that is represented by daily summaries of student clicks based on courses and by their actions.

Keywords: *machine learning, data preprocessing, ETL, multiple data sources, demographic data, assessment results, Moodle logs, Web logs*

I. INTRODUCTION

Many new pathways and insights for institutional effectiveness and the learning sciences are opening up due to the growth of information and educational technologies like Learning Management Systems (LMS), SIS. These technologies captured as 'digital breadcrumbs' from sources such as personality profiles, learning outcomes and behaviors; hence various analytics systems are emerged for improvement in institutional decision making, advancements in learning outcomes for at-risk students, significant evolutions in pedagogy more accurately and easily. Over time, their digital records may be augmented with other information, including financial and awards, involvement on campus, disciplinary and criminal reports, and personal health information. This increasing amount of admitted student data available on various data sources, the new technologies for linking data across datasets, and the increasing challenges need to integrate structured and unstructured data are all driving new aspects. Then, data collection and preprocessing are the most important and essential stages to acquire the fine and final data from multiple data sources that can be lead to correct and suitable for further data mining tasks.

By seeing current student data situation, data preprocessing becomes more crucial part and plays as a key concept of a system .There are a number of data preprocessing techniques rely on requirements and features of system's data model. Since last decade, ETL process became fruitful to flow preprocessing step smoothly. After data preprocessing, consistent data will be pulled from multiple sources and loaded to data warehouse. After retrieving some meaningful and knowledge information, the system needed to do for understanding and identifying the correlation rules between student assessment, behavior and personality traits with machine learning algorithm. There are many aspects to achieve in academic system. But the main idea of the proposed research work is that clustered analysis result are firstly explored based on students' academic performance and behavior of each student. After that, with their personality results from online survey test are merged to conduct the correlated rules between them.

The introduction of today's educational technologies situation and advantages, ETL preprocessing with multiple sources are presented in this section. The remainder of this paper has been arranged in different sections. Section 2 describes the related researches in this area. Then section 3 implements terminology of ETL. Implementation and

Experiments will be presented in section 4. Conclusion is made in the final section.

II. Related Work

In recent past, most of the related works are implementation of predicting academic achievement based on personality model and then focusing on students' previous marks, other historical data and students' behavior analysis by using data mining techniques approaches. Due to the development of educational technologies, there are many statistics and evolution settings to provide better academic environment by tracking, aggregating and analyzing student profiles along with all of the digitalized data of students.

Some of the research works with ETL processing in higher education are reviewed in this section. There are various implements in the processing of data in ETL process to suit with the requirements of the system. To qualify for work in academic data analysis model with current data situation, building preprocessing step is one of the challenges and important parts of a model. In [1], the authors evaluated student behavior clustering method based on campus big data with density based clustering method which is parallelized on the Spark platform and applied to subdivide student behavior into different group using historical data are as source in digital campus shared database. This proposed approach is to study for universities to know students well and manage them reasonably and improved algorithm is also effective. And then, the authors proposed ETL as data acquisition and preprocessing by integrating multi-source data before loading to the target system. Because this has been the way to process large volumes of data that it can scale cost effectively.

In [2], the authors explored the framework of the modern data warehouse with big data technology to support decision making process in academic information system. To reduce difficulties associated with traditional data warehouse, they designed a decision support system for big data by involving Hadoop technology. They also discussed ETL architecture based on the characteristics of the traditional data warehouse technology which cannot handle unstructured data and modern data warehouse. In [3], the authors proposed a framework for development of flexible educational data mining application to facilitate self-discovery of rules and trends from educators with little technical skills to various user types. This framework is also demonstrated with utilizing tools which are used for data mining analysis within a LMS. The authors used ETL stage as preprocessing for extracting the information from the LMS or e-learning system to do analysis much easier.

In [4], the authors implemented the case study in business intelligence framework by using data integration and ETL and then described its significance for better higher education management. They detected the functions of data integration and ETL tools and described how the correlated to each other to develop business intelligence. Because of data integration is one of the important components and ETL is the common steps to integrate data and transform it to targeted system from different academic data sources for higher education. They demonstrated on the Graduate Studies Management System (GSMS) database of University Technologies of Malaysia to store basic information of student. To ensure the data is reliable and could make decision accurately with ETL process, this database must be integrated with various sources which have different platform and format.

III. ETL Terminology

There are solutions coming up for better data integration and data warehousing because of data management has been evolving rapidly. ETL is a popular architectural pattern and used for data process with necessary integration from heterogeneous and distributed data sources with different format. ETL procedures is needed to dedicate based on the design and implementation of the system because designed ETL process are expensive to maintain, alter, and upgrade, so it is crucial to make the right choices in terms the best innovation will certainly be used for developing and preserving the ETL procedures.

A. Data Warehouse and ETL

Data Warehouses are used in higher education for decision making to take an action and to predict risk and opportunities of students. It is a process of collecting and managing data from varied sources to achieve reliable, accurate and meaningful information from various data sources. There are three steps to follow before storing data in a data warehouse, which is called ETL. This three-step process takes place when information from one system needs to be moved into a data warehouse environment. Data extraction involves extracting data from homogeneous or heterogeneous sources; data transformation processes data by data cleansing and transforming them into a

proper storage format/structure for the purposes of querying and analysis; finally, data loading describes the insertion of data into the final target database such as an operational data store, a data mart, data lake or a data warehouse.

B. ETL with Big Data

The need for ETL has increased considerably due to modern data analytics operations often have to process with rise in data volumes as quickly as possible. Therefore traditional ETL approach has to develop to function of more constrained data storage and data availability because it can slow down the process significantly for systems optimization. Apache Hadoop provides a cost-effective and massively scalable platform for ingesting big data and preparing it for analysis [5]. Using Hadoop to offload the traditional ETL processes can reduce time to analysis by hours or even days.

Moreover, ETL on platform like Hadoop and Spark give ETL a new look because it changes the cost structure around harnessing big data and save time too. [6] From their study, they recognized each ETL process instance handles a partition of data source in parallel way to improve further ETL performance facing the big data by using parallel/distributed ETL approach (Big-ETL). The researchers developed ETL functionalities which can be run easily on a cluster of computers with Map Reduce (MR) paradigm. Apparently, by using this approach on very large data integration in a data warehouse to qualify in good performance of ETL process significantly.

IV. Implementation and Experiment

In today's education world, for better or worse, testing and data are driving decision more and more as universities seek to be evidence-driven by using all of student related data. To fulfill their needs and know their weakness in time for their educational life, there are various new trends and aspects in need to collect, analyze, interpret, store, track, aggregate the increasing amount of admitted student data available from various data sources.

Apparently, there need to explore accurate and effective way of data collection and preprocessing for linking data across datasets from multiple data sources. Accurately, the proposed system collects student related data from three different data sources by using ETL as preprocessing stage for the collection analysis of students' data. Demographic data and assessment results are collected from SIS. It is very time saving and effective way to observe all students' related data and many suggestions and ideas can get from these data which are used to investigate the academic growth. Then logs of their interaction with Moodle are also collected which is one of the best data sources to know about students' behaviors based on their interested course and relevant teachers. To know their academic behavior more accurately, aggregating with Web logs which are represented by daily summaries of student clicks based on courses and by their actions. In above Fig 1, overview of proposed system for academic data analysis with ETL preprocessing is described in details. For research environment, University of Computer Studies Yangon (UCSY) is used for students' datasets as sample.

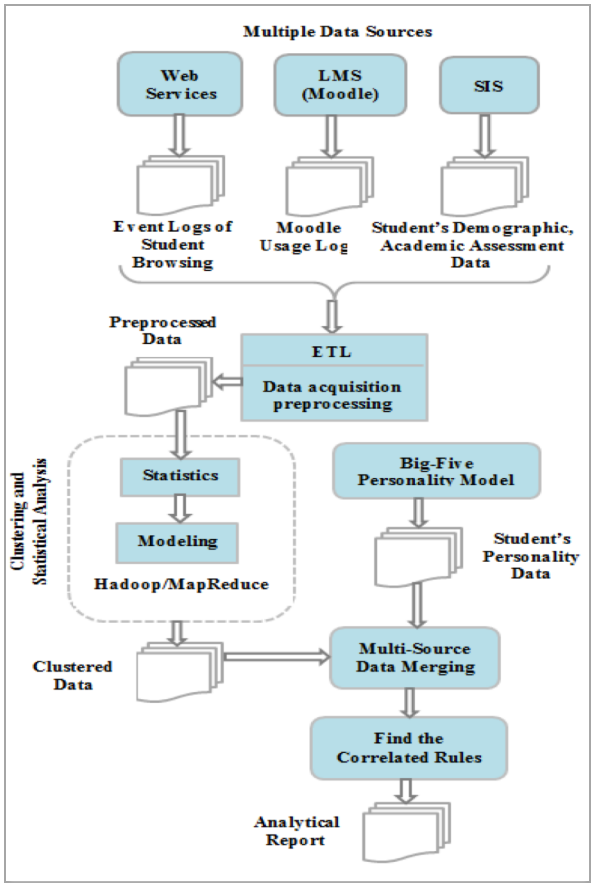

Figure 1. Proprosed System of Academic Data Analysis with ETL Preprocessing

A. Data Collection and Implementation of ETL

As in fact of the data processing of the system, each student represents the portion of data which are correlated and these are from multiple data sources of the university. Therefore, building ETL preprocessing based on the proposed system's used data scale detect a formal representation model for capturing the ETL

process that map the incoming data from different data sources to be in a suitable format for loading to the target system. In this ETL process, the system prepares the data model for building datasets of machine learning algorithm. Therefore, all of the columns are not processed in transaction data.

Although many other features are existed, the system defines some of attributes which are more dominant for data processing of the system especially helps to know an individual in academic life. In the extract phase of ETL process, the system will store the transaction data in staging. Based on the demographic data as a sample, only *EnrollmentNo, RollNo, ParentalEducation and MathScore* will be loaded into the warehouse model among the attributes such as *RollNo, Name, Email, SectionID, EnrollmentNo, NRC, Address, FatherName, ParentalEducation and MathScore*. *EnrollmentNo* is taken as unique key for the students and *RollNo* will be taken to link with other tables. Thereafter, anonymization is necessary for *EnrollmentNo* due to student information privacy. According to the ethical and privacy requirements, this system will also add annonimization for the data privacy. Experiment of ETL process with sample datasets is conducted on Python. After applying hash function to *EnrollmentNo* of some student, the forms of anonymized data are emerged as in below Fig 2:

[-485071565423000, 6636358586980677504, 5465913911662764385, -7622275796521486195,

Figure 2. Example of Anonymized Data with Hash Method

The sample assessment datasets are received by course code. These datasets are extracted by each course which is described in Fig 3. From this data, attendance and other assessment results of each student are needed to compute in sum up and score by *RollNo.* Then these computed scores are to be taken into database which is also described in Fig 4.

	RollNo	SectionID	Semester	Tutorial	Attendance	Total Assessment	Exam
0	1CS - 1	A	First	8	8.0	16.0	20.0
1	1CS - 2	A	First	7	3.0	10.0	25.0
2	1CS - 3	A	First	9	10.0	19.0	20.0
3	1CS - 4	A	First	8	3.0	11.0	16.0
4	1CS - 5	A	First	10	10.0	20.0	18.0

Figure 3. Example of Students' Assessment Data by Course

	Attendance	Total Assessment	Exam
count	316.000000	316.000000	316.000000
mean	6.933544	13.892405	15.933544
std	2.193500	4.443058	5.605441
min	1.000000	1.000000	0.000000
25%	6.000000	12.000000	11.000000
50%	7.000000	15.000000	17.000000
75%	9.000000	17.000000	20.000000
max	10.000000	20.000000	29.000000

Figure 4. Computing score of a Student by Course

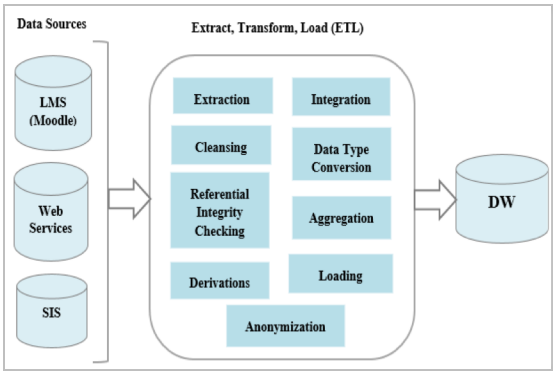

Figure 5. Steps within ETL Process

In transformation phase is cleaninging and confirmation steps of ETL. Its purpose is to gain accurate data which are correct, complete, unambiguous and consistent. In loading phase, warehouse model of this system is not like traditional warehouse and its purpose is to build the dataset for model building in analytical process. After that, the aggregation model (Data Warehouse) is explored but we need to change data model to suit for Machine Learning Algorithm. In Fig 5, ETL processing steps of the system is also described.

B. Density-Based k-means Clustering Method

After the system proposed ETL as data acquisition and preprocessing by integrating multi-source data, the system evaluates the cluster analysis results as a performance of students in different groups from students data along with their academic digitized record by using density-based k-means clustering method as improved k-means clustering algorithm on the Hadoop platform and applied to subdivide student academic performance and behavior into different group.

Then, the system extracts the valuable attributes that reflect in evaluation of cluster analysis results. With traditional k-means clustering and improved k-means clustering algorithm, the system analyses the statistical results and compare the accuracy between them. To provide more accurate implementation

method, k-means algorithm based on density partitioning is needed. In fact, the system construct initial clustering center set based on density not k value. Thereafter improved k-means clustering algorithm is developed by applying the selected initial clustering center. The method first determines a similarity measure method suitable for student data to normalize data and then builds initial clustering center set based on the density.

The density of the data object X_i in the sample set $S=\{X_1, X_2, \ldots, X_n \mid X_i \in R^t\}$ is defined as the number of samples within the ***Eps*** neighborhood of X_i. The density reflects the intensity of the sample points in the neighborhood. The density threshold ***minPts*** is the specified division of the core and the isolated point of the density range, which can be artificially set. The density parameter $N_{Eps}(X_i)$ of the sample is calculated as in 1:

$$N_{Eps}(X_i)=\{X_j \in S \mid 0 \le D(X_i, X_j) \le Eps, j=1,2,\ldots,n\} \quad (1)$$

Where $D(X_i, X_j)$ represents the distance between two samples in S. Actually, k-means doesn't allow development of an optimal set of clusters and so for effective results, the need to decide on the clusters before. Then, the statistic of traditional k-means clustering is difficult to predict k-value, different initial partitions can result in different final clusters.

C. The Big-Five Personality Test

Finally, the students' personality results and the clustered results are merged with their student identification number and then analyze with Apriori association rule algorithm, the most classical and important algorithm for mining frequent item sets, to conduct the correlated rules between them which are firmly and consistently associated. In this system, 50-questions (items) inventory is used to measures as individual on the Big-Five personality dimension that is recreated from the Big-Five Inventory (BFI). To achieve students' personality results, they need to take online personality survey test from Moodle.

The development of questionnaires measuring the Big-Five personality traits is common in psychology research for different reasons. There are countless personality tests designed with many formats. But the Big-Five personality model is one of the popular models and it also called as the Five-Factor Model (FFM). The personality survey test is designed to assess the Big-Five factors of personality: *Extraversion (E), Agreeableness (A), Conscientiousness (C), Neuroticism (N), and Openness to Experience (O)* [7,8]. In the system, the items on the BFI scale were also scored with a *Likert-scale* (5-point) from "strongly disagree" to "strongly agree" and each factor represents ten questions respectively. Some example phases are:

- *I am talkative. (E)*
- *I feel little concern for others. (A)*
- *I am always prepared. (C)*
- *I get stressed out easily. (N)*
- *I am an inventive. (O)*

Each personality trait is associated with a set of statements and the score of a trait is calculated for each user on the associated questions. After data collected through the questionnaires, Statistical Package for the Social Sciences (SPSS) is used for statistical analysis. Firstly, the system need to test the reliability and validity of the questionnaire in data collection by reason of the results of research quality. In addition, reliability estimates conduct the amount of measurement error and validity determines the questionnaire compiled it valid or not in a test. In this paper, 30-participants are only participated as sample to test the reliability and validity of system's personality survey test questionnaires.

Cronbach's alpha is the most widely used objective measure of reliability or internal consistency and it is a simple way to measure whether or not a score is reliable. But in some questions often contain some items with negative sense which are need to be reversed score before run Cronbach's alpha in SPSS. Reverse scoring means that the numerical scoring scale runs in the opposite direction and so the system need to reverse-score all negatively-keyed items. For interpreting alpha for *Likert-scale* question: 0.70 and above is good, 0.80 and above is better, and 0.90 and above is best [9].

In Table 1, the reliability of the questionnaire (30-participants to complete the N= 50-item) in data collection are tested with compare result based on reverse-score are described. After reverse-score the negative worded questions, the scale had an internal consistency is α=.76. However, a high coefficient alpha does not always mean a high degree of internal consistency and it is also affected by the length of the test.

Table 1. Comparison of Reliability Statistics

	Cronbach's Alpha	Cronbach's Alpha Based on Standardized Items	N of Items
Without Reverse -Score	.588	.490	50
With Reverse-Score	.762	.756	50

As pointed out earlier, test the validity of the questionnaire is conducted using Pearson Product Moment Correlation (PPMC) using SPSS. It is the test statistics to measures the statistical relationship, or association, between the continuous variables and based on the method of covariance. This validity test is done by correlating each item questionnaire scores with the totally score. The sign of the correlation coefficient r indicates the direction of the relationship, while the magnitude of the correlation indicates the strength of the relationship in the range -1 to 1 [10]. In Fig 6, the validity of the questionnaire (N=30-participants with sample questions of Extraversion) in data collection are tested.

		I am talkative.	I start conversations.
I am talkative.	Pearson Correlation	1	.790**
	Sig. (2-tailed)		.000
	N	30	30
I start conversations.	Pearson Correlation	.790**	1
	Sig. (2-tailed)	.000	
	N	30	30

**. Correlation is significant at the 0.01 level (2-tailed).

Figure 6. Pearson's Correlation among the Items

In fact of above figure, Pearson's correlation of two questions is (r=0.790) which indicate that the strength of association between the variables is very high. Then Sig(2-tailed) value is 0.000<0.05. This means there is a statistically significant correlations between two variables. After the testing of reliability and validity, with the factor analysis on scale items, each student has five traits which are scored on a continuum from high to low. According to calculated result, score description of each trait is as an example in extraversion, high scores tend to be very social while low scores prefer to work on their projects alone.

Finally, the main idea of the proposed approach is explored by merging based on students' identification number to find the correlations between the students' personality results and the clustered results. Then to conduct the correlated rules between them which are firmly and consistently associated by analyzing with Apriori association rule algorithm. Therefore, from the final resulted correlated rules, students' academic performance and behaviors are correlated or not on their personality results.

V. CONCLUSION

Higher education is working in a more and more complex and there need to explore accurate and effective way of data collection and preprocessing for linking data across datasets from multiple data sources. This paper described on the ETL preprocessing step for the collection and analysis of student academic digital record from multiple data sources to ensure the data is reliable and could contribute to decision making for work in academic data analysis model. Then, to qualify for the results of research quality in data collection, a part of measuring the personality test questionnaire are also described in this paper.

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The Implementation of Support Vector Machines for Solving in Oil Wells

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Abstract

The article deals with the problem of timely forecasting and classification of problems that arise in the process of well construction remains relevant. It is necessary to create a new methodology that should help drilling personnel to make timely decisions about possible problems in the drilling process on the basis of real-time data analysis, which will increase efficiency and reduce drilling costs accordingly.

Keywords*: drilling complications, machine learning, neural network, efficiency improvement, gradient boosting, classification*

I. INTRODUCTION

With the development of oil well digitization, both the data source for mass production parameters and the real-time data collection method support oil production with an optimized solution [1]. Using machine learning to improve, combine, modify, improve applications, and optimize oil well data analysis is a new smart scientific method of the oil well data analysis system. Currently, the parameters of an oil well used in the data analysis algorithm are relatively simple, in the absence of polyphyletic parameters, a standard for evaluating and terminating data [2]. In addition, in some oil wells that have entered the middle or later periods of the high water stage, features such as low permeability and resistivity of the complex accumulation layer may cause the General manual analysis and linear analysis to be invalid [3]. From the perspective of intelligent machine learning, a nonlinear SVM classification algorithm is proposed in this paper, the structure of the data development system and the pattern recognition model for polyphyletic parameters are constructed, and the use of SVM through a high-dimensional spatial feature map and hyperplane optimized classification allows solving the problem of analysing nonlinear parameters of oil wells and pattern recognition.

II. PATTERN RECOGNITION OF OIL WELLS

In the course of oil production, the monitoring centre collects, transmits, analyses and provides real-time data on the flow rate of oil and gas for oil production, product watering, pressure, temperature, electrical voltage, electric current and load, as well as other primary parameters, which helps the administrator understand the operating conditions of the oil well and ensure its operation in a high-efficiency and low operating flow mode [4]. As a rule, these parameters also include peak values of electric current and voltage, pump pressure, back pressure, oil pressure and pressure in the annular space of the well.

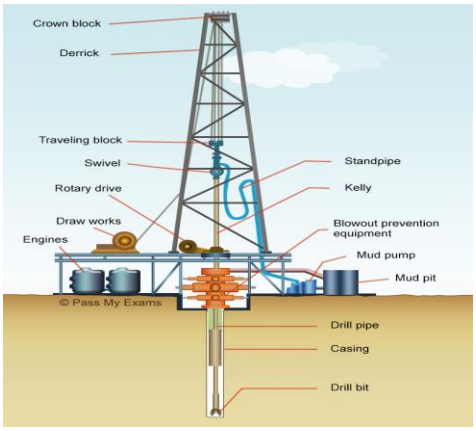

Figure 1. Intelligent systems in oil fields

This data is transmitted to the automated control system in real time. After performing a linear approximation and forecasting of the obtained data, the decision-maker can assess the state of the well at the moment and predict its behaviour in the future, and take appropriate compensating control actions.

III. NONLINEAR SVM

The kernel method allows for solving the problem of nonlinear classification using a nonlinear transformation [7]. Provided that the input space is Euclidean and the feature space is Hilbert, the kernel method means the product of feature vectors obtained by converting input data from the input space to the feature space. Using the kernel method to study nonlinear data in order to obtain a nonlinear SVM. The entire procedure is the operation of the linear SVM method in a multidimensional feature space.

The General idea is to use a nonlinear transformation to change the input space into a feature space, which can transform the hypersurface model in the source space into a hyperplane in the feature space. This means that a nonlinear classification problem in the original space is transformed into a problem that can be solved b y a linear SVM in the feature space [5].

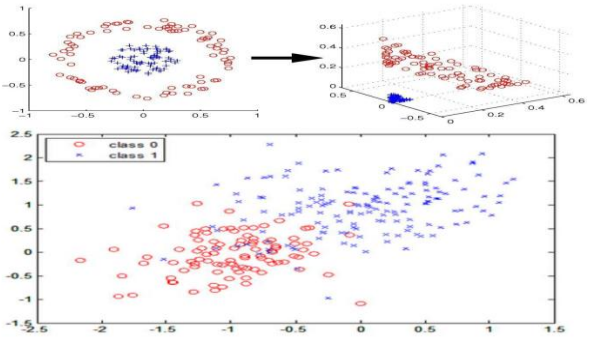

Figure 2. Using the kernel method to solve a nonlinear problem

IV. SUPPORT VECTOR MACHINE

The SVM model builds a hyperplane or set of hyperplanes in a multidimensional space called the feature space, which can be used for classification or regression. Its advantages over other machine learning methods include greater generalization capability, strong noise immunity, and less learning time (Vapnik V. 1995; Ani-fowose and Abdulraheem 2011). The SVM approach was developed in 1992 by a company Vanpik in collaboration with the Laboratory of Bell Laboratories. The SVM model is a set of interrelated managed learning methods used for classification and regression [6]. The SVM principle is based on statistical learning theory and structural minimization, which has shown better performance than the usual empirical risk minimization used by many machine learning methods (James Lara 2008).

$$min_{w,b,\varepsilon} \frac{1}{2}||w||^2 + C\sum_{i=1}^{N}\varepsilon_i. \quad (1)$$

$$\text{s.t. } y_i(wx_i + b) \geq 1 - \varepsilon_i \quad (2)$$

Where C is the penalty parameter. Increasing C also increases the penalty for classification errors. You must adjust the target function to minimize the number of singular points while maximizing the offset from the hyper plane.

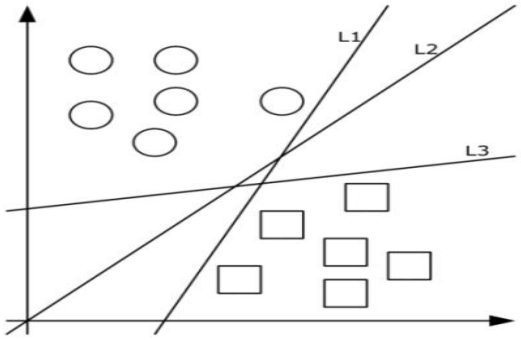

Figure 3. Hyperplanes and Support Vectors

V. LINEAR LOGISTIC REGRESSION GORITHM

The linear logistic regression algorithm is a classic classification method for studying statistics related to the linear logarithmic model [8]. This classification model is a conditional probability distribution P (Y / X), which is a judgment model. It can be obtained from the linear regression model hw (x)= $_wT_X$ and the sigmoid curve:

$$P(Y = 1|X) = \frac{1}{1+e^{-wx}}. \quad (3)$$

Where X is the input, Y is the output, W is the weighted coefficient, and WX is the internal product. The logistic regression distribution function and the density function are shown in Fig. 3. Logistic regression compares the difference between two conditional probabilities and classifies the training example x

into a large probability group. For the data of the training set it is possible to use maximum likelihood to estimate the parameters of the model to obtain the logistic model. The following assumptions are introduced [9].

$$P(Y = 1x) = f(x), P(Y\,0 = x)\,1 = f(x) \quad (4)$$

Likelihood function

$$\prod_{i=1}^{N} [f(x_i)]^{y_i} [1 - f(x)]^{1-y_i}. \quad (5)$$

Logarithm likelihood function

$$L(w) = \sum_{i=1}^{N} [y_i \log f(x_i) + (1 - y_i) \log(1 - f(x_i))]. \quad (6)$$

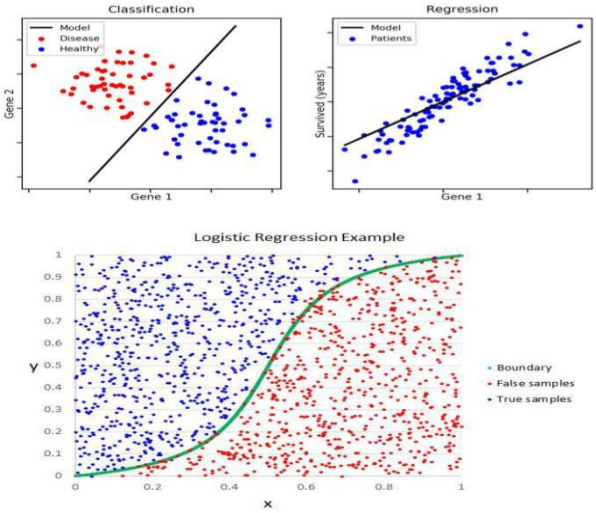

Figure 4. Logistic regression distribution function and density function

VI. IMPLEMENTATION AND RESULTS OF THE EXPERIMENT

The main purpose of the study is to evaluate the effectiveness of oil well planning. The efficiency of the system is the most important factor in the quality of the production system. The efficiency of a production system is the ratio of the useful amount of liquid produced to the power consumed per unit of time, which is a significant factor in production. As a result of the experiment, the efficiency of the system was chosen as the target factor. It is assumed that the system efficiency value above 45% is positive, less than 45% is negative.

In data mining, parameters such as pump load, temperature, and electrical voltage are suitable for solving the classification problem in the evaluation model. When analyzing the efficiency of the pumping system, the factors that affect it listed in the table are taken into account.1 and table. 2 [10].

Table 1. Oil well parameters

Parameter	Unit	Parameter	Unit
Reactive power	[KW]	Depth	[m]
Oil pressure	[MPa]	Period of work	[/]
Max pressure	[MPa]	Max load	[KN]
Min pressure	[MPa]	Min load	[KN]
Power factor	[1]	Production pressure	[MPa]
Voltage	[V]	Active power	[KW]
Current	[A]	Max active power	[KW]

Table 2. Oil well production parameters

Parameter	Unit	Parameter	Unit
Doppler velocity (array)	[Hz]	Liquid Consumption	[m^3/day]
Gas void fraction (array)	[%]	Gas Consumption	[m^3/day]
Sound velocity	[m/s]	Water cut	[%]
Fluid Pressure	[MPa]	Temperature	[℃]

Primary data were obtained for oil fields in the Perm region, for oil wells and booster pumping stations for a long period of their operation [11].

The first block of data presented in table 1 was relatively easy to obtain, since these parameters are measured directly by the corresponding sensors. The data shown in table 2 are the results of measurements of an innovative ultrasonic multiphase flow meter. It was installed in oil wells and booster pumping stations and consists of a vertical measuring tube with two different calibrated sections, four Doppler sensors, four gas void fraction (GWF) sensors, two sound velocity sensors, a thermometer, a sensor, and a computing unit with a multi-layer mathematical model. This model sets the boundary between the primary data (Doppler speed, GWF, speed of sound, temperature, pressure) and the calculated data (liquid flow, gas flow, and water flow).

The main parameters of oil production efficiency are the flow rate of liquid, gas and water content. However, using the primary parameters, you can determine the flow mode of the mixture. There are four main types of flow modes: bubble, mucus, dispersed ring, and dispersed [12]. The flow mode shows the stability of the oil well operation, and it should also be taken into account when evaluating the efficiency [13].

To reduce the feature space, the integral values of the Doppler velocity and GWF can be calculated as the arithmetic mean of the four corresponding parameters. Parameters are measured at four points of two different calibrated cross sections of the pipe: in the center of the small cross section, on the periphery of the small cross section, in the center of the large cross section, on the periphery of the large cross section [14].

According to the mathematical model of the flow meter, the flow rate depends on the primary data of the twelve parameters. However, as an example in Fig.1 shows the relationship between the fluid flow rate and the integral Doppler velocity.

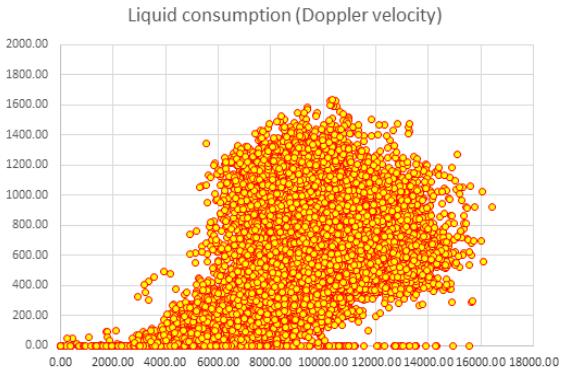

Figure 4. Dependence between liquid consumption and integral Doppler velocity

Figure 4 shows that when the Doppler velocity increases, the fluid flow increases. However, fluid flow also correlates with other parameters such as GVF, water flow, temperature, pressure, and others. And the main correlation between the flow rate of the liquid and the integral Doppler velocity is blurred under the influence of these parameters [15].

For rice. 2 shows the relationship between the gas flow rate and the integral fraction of gas voids.

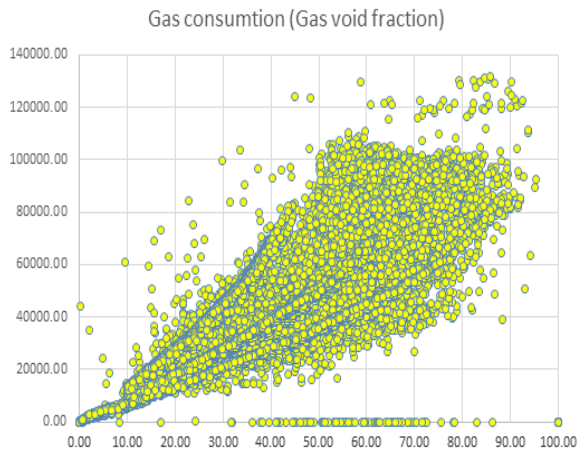

Figure 5. Dependence between gas consumption and integral gas void fraction

It follows from the figure 5, it is shown that the higher the GWF, the greater the gas consumption. However, gas consumption also depends on other parameters, such as fluid velocity, water content, temperature, pressure, and so on. the main correlation between gas consumption and GWF is also blurred by these parameters. Moreover, the greater the absolute value of the GWF, the greater the variation in gas flow [17].

Therefore, after analyzing cross-correlations in the training set, it was determined that all parameters should be used in the study.

To improve the results of the work performed in accordance with the received data, the following actions were performed.

1) to improve the efficiency of data collection, all possible related information was collected.
2) created an evaluation model for solving real problems.
3) an optimal modular scheme has been developed.
4) results are compared with real data and the model is updated.
5) the data set was pre-processed using anti-aliasing, normalization, and noise reduction techniques.
6) the evaluation of the obtained models was carried out.

VII. Classification Results

As a training set, 2019 examples of data from oil wells and pumping stations were selected, and 20 examples were selected as a test sample. According to experience, the penalty parameter C was set to 0.8, the RBF estimation function and the standard deviation 0.5 for the SVM model; the penalty parameter C=1 for LR. A comparison of projected and actual

performance indicators is shown in table 3. The experiment was performed in python on the example of oil well data using SVM and LR algorithms. Five years of measurement experience and primary and calculated data were obtained, where they were systematized and cleared.

№	Real values	Forecast LR	Forecast SVM	№	Real values	Forecast LR	Forecast SVM
1	0	1	0	11	0	0	1
2	0	0	0	12	0	0	0
3	0	0	0	13	0	0	0
4	0	0	0	14	0	0	0
5	0	0	0	15	0	0	0
6	0	1	1	16	1	1	1
7	1	0	1	17	0	1	0
8	1	1	1	18	0	0	0
9	0	1	1	19	0	0	0
10	1	1	1	20	1	1	1

Using the logistics model, 18 correct classifications were found with 90% accuracy that meet the forecast conditions. Using the PCA dimension reduction method, the data dimension was reduced from 17 to 2. The deviation of the result from the graph of real values is shown in Fig.6. 15 correct classification results were found, which means that the accuracy reaches 75%. Within the SVM model.

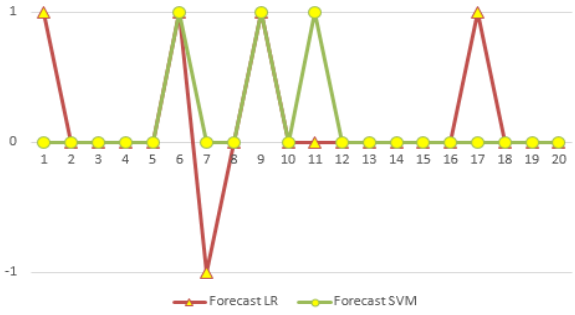

Figure 6. Results of the SVM and LR experiment

In the subject area of oil wells, data distribution is complicated by the high dimension of the information space, which can have a large impact on the collection of primary data. In this situation, there may be errors in the collection of one or more data types, as well as uneven distribution of data. Classic manual analysis, such as using charts, linear analysis, or logistic regression, does not provide qualitative classification. In this case, a support vector machine using the kernel method is better suited for nonlinear complex data processing. In this case, the current operating mode of oil wells is considered automatically based on a set of primary data that leads to lower or higher values of liquid and gas flow rates and to lower or higher efficiency values, without direct classification of oil wells by this parameter. However, it would be very useful to develop such a classification for predicting the behavior of oil wells and preventing accidents. For figure 6 deviation ratio between the real value and the forecast of LR is shown by triangles, and the deviation ratio between the real value and prediction of SVM are shown by circles.

VIII. Conclusion

The article presents a theoretical analysis of the support vector machine and logistic regression. It is shown that the nonlinear SVM algorithm works better than the linear LR algorithm when analyzing the oil well system and predicting its efficiency. In future studies based on the current review, it is necessary to develop a method of multiple classifications based on the support vector machine, which allows you to classify the original data set into several classes with the ability to assess the degree of proximity to each of these classes. For an oil field process service, it is very important to determine the multiphase flow mode (oil-water-gas), since, for example, even a high flow rate in an inappropriate mode can lead to a pump failure and an emergency stop of the oil well. However, if this situation could be classified at an early stage, it would avoid an emergency situation and thus preserve the efficiency of oil well operation.

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Geographic Information Systems
and
Image Processing

An Efficient Tumor Segmentation of MRI Brain Images Using Thresholding and Morphology Operation

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Abstract

In medical image processing, segmentation of the internal structure of brain is the fundamental task. The precise segmentation of brain tumor has great impact on diagnosis, monitoring, treatment planning for patients. Various segmentation techniques are widely used for brain Magnetic Resonance Imaging (MRI). The aim of this paper presents an efficient method of brain tumor segmentation. Morphological operation, pixel extraction threshold based segmentation and Gaussian high pass filter techniques are used in this paper. Thresholding is the simplest approach to separate object from the background, and it is an efficient technique in medical image segmentation. Morphology operation can be used to extract region of brain tumor. This system converts the RGB image to gray scale image and removes the noise by using Gaussian high pass filter. Gaussian high pass filter produced sharpen image and that improves the contrast between bright and dark pixels. This method will help physicians to identify the brain tumor before performing the surgery.

Keywords: *Image segmentation, Thresholding, Morphology operation, Preprocessing*

I. INTRODUCTION

Various segmentation techniques are used in medical images processing for different accuracy and degree of complexity. Image segmentation is the method of separating an image into different regions which is performed to detect, extract and characterize the anatomical structure of brain. Tumor is one of the most dangerous diseases in any part of the body, and has different characteristics and different types. The tumor occurs in fatal parts of the body in which total system is dysfunctional, when it is in the brain [1]. Brain tumor is an abnormal mass of tissue in brain and is classified based on the involved tissue type, tumor location whether it is benign or malignant, and other factors. Abnormal tissues will make more mistakes for brain area. Although many efforts and favorable results in the medical imaging community, accurate and performance segmentation and characterization of abnormalities are still a challenging and difficult task [2]. MRI segmentation is an imaging technique used to provide invaluable information about anatomical structure of brain. This paper presents an efficient method based on thresholding and morphology operation for brain tumor segmentation using MRI images. Medical images have very complex distribution of intensity so segmentation using thresholding often fails. However, thresholding methods are often combined with other segmentation methods. Thresholding is an efficient method for image segmentation because it reduce computational time, and faster than other method. But the spatial characteristics of an image do not work into account [3].

Brain tumor segmentation would be difficult to solve because it usually involves an enormous data. These facts are the most challenging and difficult in many medical images. Brain involved many types of tumor which have various shape and size. In this paper, accurate segmentation of brain tumors are identify in MRI images. In first stage is pre-processing operations. The second stage is brain tumor segmentation using thresholding. And then extract the tumor in segmented image using morphology operation.

Nowadays, many segmentation techniques can be considered using manual segmentation method. This method will become an error-prone and time consuming task for the expert. Brain MRI segmentation can help the radiologists and clinicians, who they decide accurately the patient's condition. Moreover it can effectively not only physicians but also patients. In fact, physician can identify the brain tumor before performing surgeries. Furthermore, patients can get accurate treatment from physicians by using segmentation of MRI imaging. The remainder of the paper is organized as follow. The next section is the concern of the brain tumor system. In section 3, we

discuss the preprocessing step of the system. In section 4, addresses the metrologies of the system. In section 5, discuss the proposed system. In section 6, addresses the experimental results of the system. Finally section 7 presents the conclusion.

II. Related Works

Abnormality, surgical planning and post-surgery are significant tasks in medical application.

Dey, et al.in [4] addressed robust watershed segmentation of noisy image using wavelet. This paper describes that using soft threshold wavelet on the region based watershed segmentation on noise image contribute a very effective result.

Alpana, et al.in [5] presented an effective method based on convolutional neural networks (CNN). Preprocessing used median filtering to remove the noise. Finally the CNN is used to identify brain tumor using MRI images.

Pooja Thakur et al.in [6] discussed brain tumor segmentation and detection. These methods are based on watershed segmentation and morphological operation. This paper described very good segmentation results, and reduced the computational time by using watershed segmentation. Furthermore, it detected the brain tumor. This paper will help the physician to surgery because watershed segmentation and morphological operation can calculate the area of brain tumor.

Ivana Despotovi et al.in [7] discussed image segmentation in medical application. The important steps of image segmentation are fundamental concepts of MRI segmentation of the human brain and basic concept of MRI segmentation methods. Image definition involves two dimensions and three dimensions of image. Furthermore image features and brain MRI intensity distributions. Then this paper reviewed preprocessing steps and the most popular image segmentation methods.

ARCHANA CHAUDHARI et al.in [9] proposed seed selection method. These method discuss using region growing algorithm. Automatic seed selection in segmented tumor is used fuzzy c- mean algorithm from MR brain images.

Jin Liu et al.in [8] described brain tumor segmentation in MRI images. Segmenation methods describe overview. Authors discussed the preprocessing of MRI images. Furthermore, brain tumor segmentation methods are discussed. There are conventional methods, thresholding-based methods, region- based methods, classification and clustering methods. In this method, some algorithm used to implement the development.

Shilpa Kamdi et al.in [9], discussed image segmentation and region growing algorithm. This paper described three methods: threshold-based, edge-based and region- based. Region growing method is better result over conventional segmentation method. This algorithm is determine with regard to noise.

III. Pre-processing

Pre-processing has to be solved before any segmentation operations which are directly related to the accuracy of the segmentation results. Pre-processing operations include many methods such as de-nosing, skull stripping, intensity normalization etc. These operations directly impact the results of brain tumor segmentation.

Image de-noising is an essential pre-processing stage for MRI images processing. Noise in MRI image makes it difficult to precisely delineate regions of interest between brain tumor and normal brain tissues [9]. Therefore enhancement and noise reduction techniques are necessary to pre-process MRI image. There are many de-noising methods for MRI image.

Median filtering is used to remove ``salt-and-pepper´ ´like noise, and could give good results. Median filtering technique calculates the median pixels within the median window, and then the selected median value gets placed at the position corresponding to the center of the median window in the output image. High pass filter produced sharpen image and that improves the contrast between bright and dark pixels.

Gaussian high pass filter is utilized to enhance the boundary of the object in the image and will blur edges and reduce contrast [6].

IV. Brain Tumor Segmentation Methods

Usually brain tumor segmentation methods are classified into three main categories such as manual, semi- automatic, and fully automatic segmentation based on the degree of needed human interaction. Fully automatic brain tumor segmentation method, determines the segmentation of brain tumor without any human interaction, may obtain better result than manual and semi-automatic segmentation. The proposed system segmented MRI images by using two methods such as thresholding and morphology operation.

A. Thresholding

Thresholding an image g_{th} (x, y) can be defined;

$$g_{th}(x,y) = \begin{cases} 0 \text{ if } f(x,y) \le \text{thres} \\ 1 \text{ if } f(x,y) > \text{thres} \end{cases},$$

Where, *thres* is threshold. Thresholding operation, input could be a gray scale image or color image and its output is a binary image. Thresholding techniques can be categories into three techniques. These are global thresholding, local thresholding and adaptive thresholding.

B. Morphology Operation

Morphology-based image processing can be performed on image, and it involves many operations. These methods are driven by operations applied by the structuring elements (SE's) on the image matrix, which use ones and zeros to perform operation based on the distributions of said ones and zeros [10]. Erosion and dilation operations are fundamental to morphology processing and these operations are defined in term of set notation. In morphology erosion, erosion of E by F, denoted $E \ominus F$, "is the set of all elements y for which $(y+c) \in E$ for every F. That erosion shrinks or thins object in binary image. Dilation of E by the structuring element F is denoted $E \oplus F$. E overlaps by at least one element. Dilation operation causes thickening of foreground areas.

V. PROPOSED SYSTEM

Image segmentation is mostly used for measuring the brain's anatomical structure, brain changes, treatment and surgical planning. Brain tumor segmentation is the process of separating abnormal tissue from normal tissue. The normal tissue is White Matter (WM), Gray Matter (GM), and Cerebrospinal Fluid (CSF). The accurate segmentation of brain tumor Magnetic Resonance Imaging (MRI) is still challenging and difficult task, so it cannot solve all the way because of the various types of tumor intensity, shapes, and location. This system is proposed to solve above the difficulty and challenging.

The proposed system includes preprocessing, segmentation, morphological operation. In preprocessing steps, first step converts the RGB images to gray scale images. Second step used Gaussian high pass filter. This filter will increase the contrast between bright and dark pixel to produce a sharping image. Segmentation is carried out by thresholding algorithm. In thresholding algorithm, the proposed` system used Otsu's thresholding method and then morphology operation is used to segment the MRI image. The number of pixels of the brain tumor segmented area is calculated using Matlab 2018a. The block diagram of the proposed system is shown in Fig. 1.

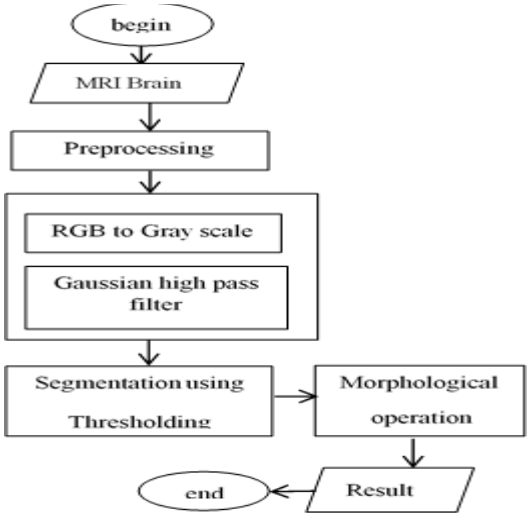


Figure 1. Block diagram of proposed system

Performance of each method would be calculated by using Peak Signal-to –Noise Ratio (PSNR) and Signal-to –Noise Ratio (SNR).

Peak Signal-to –Noise Ratio (PSNR) measures the quality of image. This quality based on pixels different between two images. PSNR is describes as

$$\text{PSNR} = 10 \log \frac{P2}{MSE}$$

Where P=255 for an 8 bit integer.

MSE= mean square error

$$\text{MSE} = \frac{1}{MN} \sum_{x=1}^{M} \sum_{y=1}^{N} \left[I(x,y) - \hat{I}(x,y) \right]^2$$

M and N are the value of row and column. x, y is current pixels position.

VI. EXPERMENTAL RESULT

The proposed system was tested with images from Internet by using 100 brain MRI images of different nature.

Figure 2(a), 3(a), 4(a), 5(a), 6(a) shows the original image of brain MRI. The original images are converted to gray scale images using Matlab as shown in Fig. 2(b), 3(b), 4(b), 5(b), and 6(b). The output of testing image is preprocessing using Gaussian high pass filter that is used to make image sharper and describes fine details in the image in Fig. 2(c), 3(c),4(c),5(c),6(c). The output of the segmentation stage is shown in Fig. 2(d),3(d),4(d),5(d),6(d). The output of the morphology operation is shown in figure

2(e), 3(e), 4(e), 5(e), 6(e). The number of white pixels (i.e., foreground regions), black pixel (i.e., background regions) and the total number of pixels are shown on the right of Fig. 2(e), 3(e), 4(e), 5(e) and 6(e).

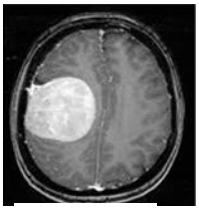


Fig. 2(a)

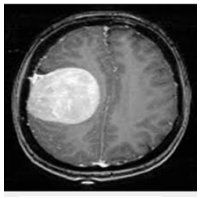


Fig. 2(b)

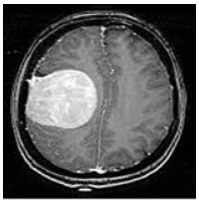


Fig. 2(c)

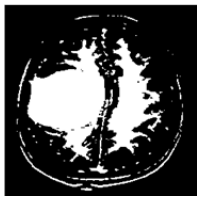


Fig. 2(d)

Fig. 2(e)

Total pixels=65536

White pixels=10606

Black pixels=54930

Fig. 2(f)

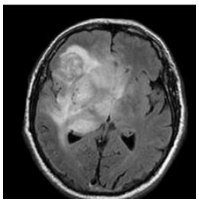
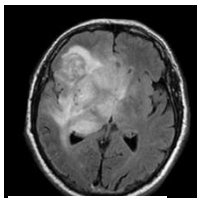


Fig. 3(a)



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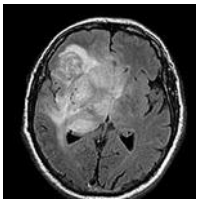
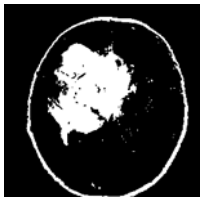


Fig. 3(c)



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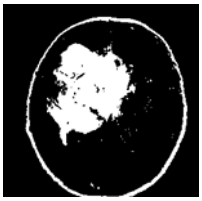


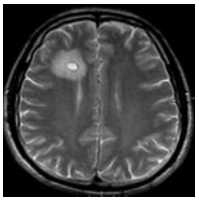
Fig. 3(e)

Total pixels=65536

White pixels=5966

Black pixels=59575

Fig. 3(f)



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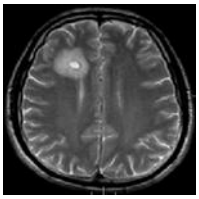
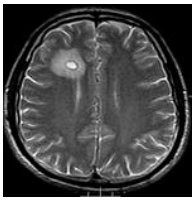


Fig. 4(b)



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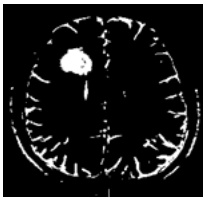


Fig. 4(d)

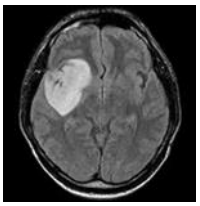
Fig. 4(e)

Total pixels=65536

White pixels=961

Black pixels=64575

Fig. 4(f)



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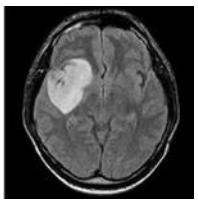

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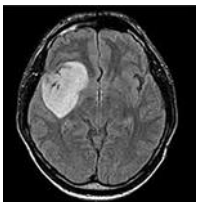

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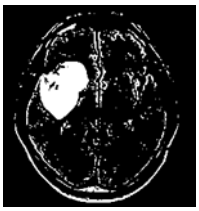

Fig. 5(d)

Fig. 5(e)

Total pixels=65536

White pixels=2239

Black pixels=63297

Fig. 5(f)

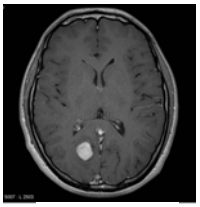

Fig. 6(a)

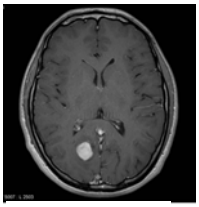

Fig. 6(b)

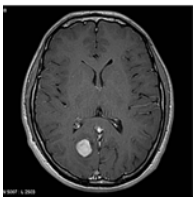

Fig. 6(c)

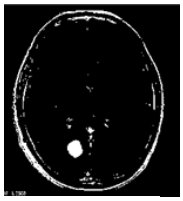

Fig. 6(d)

Fig. 6(e)

Total pixels=65536

White pixels=406

Black pixels=65130

Fig. 6(f)

The performance of this system is analyzed using Peak Signal-to-Noise Ratio and Signal -to - Noise Ratio. PSNR and SNR value of five images presented in table 1. Otsu's method and morphological operation were analyzed for segmentation. Using these two techniques, performance of difference images were measured by using PSNR and SNR. Performance evolution of proposed system analyzed by using Gaussion high pass filter. Otsu's and morphology operation do not use the Gaussion high pass filter method. An image produce better quality and sharer by using Gaussion high pass filter. Image qualities of proposed system are higher than Otsu's and morphological operation. Performance value of five images are presented in following table 1.

TABLE I. REPRESENT PSNR AND SNR VALUE OF FIVE IMAGES

Images	Proposed System		Otus's Method		Morphology Operation	
	PSNR	***SNR***	***PSNR***	***SNR***	***PSNR***	***SNR***
image2	20.3319	5.3941	19.5674	16.6256	20.2710	11.5211
image3	20.1152	5.1174	19.9301	16.7515	20.0156	9.1873
image4	19.8976	0.5580	19.3978	18.2357	19.8810	7.6215
image5	19.5674	4.9032	19.2759	16.7030	19.4284	7.6310
image6	20.1098	-9697	19.8782	17.0575	20.0045	5.7601

VII. CONCLUSION

MRI using segmentation method is an important for brain tumor. This paper has three steps, first is preprocessing of given MRI image and second is tumor segmentation and third step is extract of brain tumor. Preprocessing used RGB to convert gray scale image and Gaussian high pass filter is for improving the segmentation quality. Tumor segmentation proposes about thresholding for brain MRI and finally, morphology operation is used to extract of brain tumor. The number of total pixels, white pixels and black pixels are also calculated from the extracted brain tumor image. Brain tumor is extracted from any MRI images in this system.

Performance of each image is measured by value of PSNR. Performance of proposed system analyzed using Gaussion high pass filter, and this filter produce a sharper image and give better quality of image.

Furthermore, a large amount of computation time can be reduced in this system because Otsu's thresholding method takes less time of segment the image. Next, the area and location of extracted brain tumor will also be calculated in this system. The proposed system will be analyzed using MATLAB continuously.

ACKNOWLEDGMENT

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Real-Time Human Motion Detection, Tracking and Activity Recognition with Skeletal Model

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Abstract

Human activity recognition with 3Dskeletal model has been attracted in a lot of application area. Representations of human based on 3D perception have been occurred prevalent problems in activity recognition. In recent work with RGB-Depth cameras, expensive wearable sensors and illuminator array have been used to construct the 3D human skeleton model in recognition system. But these systems have been defined specific lightening condition, limited range. and great constraint in outdoor applications. To overcome this restriction, the proposed system is considered on the real-time video sequences of the human movement to understand human behavior in indoor and outdoor environment. The proposed method is constructed human detection and motion tracking by using framewise displacement and recognition is based on skeletal model with deep learning framework. The result is to become an efficient detection, tracking and recognition system for real-time human motion. The performance and accuracy of the system is analyzed with the various videos to show the results.

Keywords: *human recognition, skeletal model, deep learning*

I. INTRODUCTION

Real-Time human motion detection, tracking and activity recognition is one of the active research areas in computer vision and has applied in many application areas. The component of human pose can interpret visual information from the surrounding environment to real-world problem. Human activity recognition system also applies in many fields such as security, safety and human activity monitoring in many environments. Human motion analysis in computer vision system consists of face recognition, gesture recognition and body recognition. 3D skeleton-based human representations generally analyzed into four categories based on raw position, joint displacement, orientation, and combined information. Local features and skeleton based representation are useful for human representation systems [6].

Figure 1. Example result of human detection, tracking and recognition with 3D skeletal

Although 3D modeling and skeletonization system have been proposed in past decades, there are unsatisfied for outdoor application [11]. To get reliable manner, the proposed system is structured 3D skeletal model according to the component of human body parts contain 18 fundamental points defined as head, neck, shoulders, elbows, wrists, hips, knees and ankles, etc. to represent human body skeletal structure and to recognize human activity. 3D skeletal model performs us to focus hidden human body parts in 2D images and that can show people interact with each other, overlapping groups and human activity. Currently human activity recognition with 3D skeletal model has limitations in various aspects. Most of the system require the generation of data for better performance with development method. Our system is intended to perform robust 3D human skeleton system based on deep neural network that is without being

altered by different situations and environmental changes. The system is constructed a skeletal model from the perception of data through joint estimation and pose recognition. Our goal is a robust and efficient approach in human recognition system from training and testing on different data. Fig. 1 shows input video with detection, tracking and recognition result of the proposed system. The arrangement of this paper as follow: Section II concerned with about the related papers. Section III focus on detection and tracking process of human movement. Section IV shows human recognition with skeletal model. In session V, expresses the experiment and accuracy results in detail. Finally, Session VI describes the conclusion and future work.

II. Related Work

Along with several numbers of human representation approaches, most of the existing of human recognition system have been proposed with skeletal model. Hussein et al. [5] proposed human skeleton by using Covariance of 3D joints (Cov3DJ) matrix over the sub-sequences frame in a temporal hierarchy manner. That has fixed length and not depend on sequence length. They computed random joint probability distribution variable correspond to different feature maps in the region. The system deployed joint location over time and action classification with Support Vector Machine (SVM). Du et al. [8] proposed Recurrent Pose Attention Network (RPAN) that predicts the related features in human pose. The system used end to end recurrent network layers for temporal action modeling to construct a skeleton-based human representation. As the number of layers increase, the representations extracted by the subnets are hierarchically fused to figure a high-level feature to represent human in 3D space.

Plagemann et al. [1] analyzed human shape from interest points of salient human body. The system directly estimated 3D orientation vector from body part in space and learned the estimated body part locations with Bayesian network. Luvizon et al. [3] proposed a multitask framework for 2D joint and 3D pose estimation from still images and human action recognition from video sequences. They trained multi type of dataset to generate 3D predictions from 2D annotated data and proved an efficient way for action recognition based on skeleton information. Hou et al. [4] proposed shadow silhouette-based skeleton extraction (SSSE) method that analyzed silhouette information. Skeleton synthesis and 3D joint estimation is working on extracted 2D joint positions from shadow area on the ground. Major joint position is defined and result is compared with RGB-D skeletonization. Various systems with different improved methods have been occurred on skeleton based human representation.

Our system is intended to build more robust system for human motion with fast and accurate detection, tracking and activity recognition by using skeleton based system. The system flow of proposed system is shown in Fig. 2.

The proposed system proceeds as the following steps:

1) Detect human silhouette information from background for each frame.
2) Extract 2D joint projected positions.
3) Extracted 2D joint positions are categorized as sequence of body parts.
4) Generate 3D human skeleton using spatial-temporal integration of 2D joint positions.
5) Define activity recognition according to skeleton joint position.

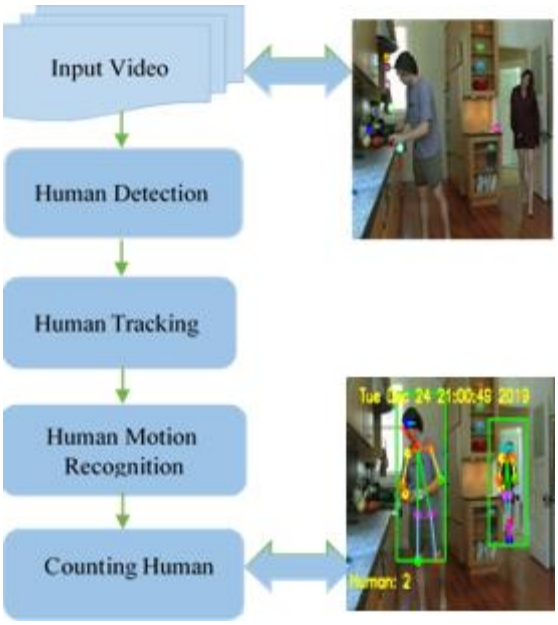


Figure 2. System flow of the proposed system

III. Human Detection

Human detection is to know human shape, pose, and motion from perception data in a frame. An efficient and accurate detection method rely on the application of human attributes. The system detects silhouette image from the background by using threshold value and the skeleton feature points is

extracted from spatial-temporal evolution of human body parts. The system finds the location of each human joint on all the feature maps [7]. And the feature map with the highest-activated value at the joint location is selected for the corresponding joint. The position of the joints is defined by using local minimum and maximum value of distance transformation and that value should be greater than their neighbor's points. The system uses iterative method on the human body until a skeleton leftover with deletion or retaining the value. The arrangement of points is based on nearest neighbor distance order according to body parts. Then, define a bounding box and detect around the human body.

A. Graph Modeling

For spatial-temporal graph modeling, the system initiates a novel adaptive dependency matrix and learns it through node embedding on connected human body parts. Our model can precisely capture the hidden spatial dependency in the data. As the number of layers increases, whose receptive field grows exponentially and handles the very long sequences for pose estimation.

B. 3D Pose Estimation

Pose estimation is an important class in action recognition. Partially, previous pose is carried as the known pose to fit and serves as a preprocessing step for next processing [2]. The estimation of 3D human pose from 2D joint locations is central to many vision problems on the analysis of people in images and video. Pose estimation can deal with any composition of rigidly moving parts connected to each other according to joints. The system identifies human body parts to estimate the pose, as in:

$$L_{pose}=\sum_I \sum_{t=1}^{T} \sum_{k=1}^{K_1\times K_2} \left(N_t^I(k)-\alpha_t^I(k)\right)^2 \qquad (1)$$

where T is total time steps, N_t^I is high visual data for all joints, α_t^I is joint attention score which can be computed, as in:

$$\alpha_t^I(k)=\frac{\exp\{\tilde{\alpha}_t^I(k)\}}{\sum_k \exp\{\tilde{\alpha}_t^I(k)\}} \qquad (2)$$

The related feature of human body parts is described, as in:

$$F_t^P=\sum_{I\in P}\sum_k \alpha_t^I(k)V_t(k) \qquad (3)$$

$V_t(k)$ is the feature vector of V_t at the k^{th} spatial location ($k=1,\ldots,K_1\times K_2$), $\tilde{\alpha}_t^I(k)$ is the unnormalized attention score of $V_t(k)$ for each joint $I\in P$, P is the body part. That describes the human figure which has relative positions of the limbs and generate 3D pose estimation that contains frontal, lateral, backwards and forwards displacement. And then tracking algorithm is applied to track the detected object across different frame.

C. Frame-wise Displacement

In Frame-wise displacement, Frame-wise Motion Fields (FMF) and Frame-wise Appearance Features (FAF) are motion representations that estimate appearance and contextual information of the video between two frames. Frame-wise displacement relative to previous frame and reference frame is computed, as in:

$$FD_i=|\Delta d_{ix}|+|\Delta d_{iy}|+|\Delta d_{iz}|+|\Delta\alpha_i|+|\Delta\beta_i|+|\Delta\gamma_i| \qquad (4)$$

Where $\Delta d_{ix}=d_{(i-1)x}-d_{ix}$, these variable measures the movement of any given frame. We have to consider the cross-entropy method to fix uncertain matches in information. This leads to very fast and reliable matching among a large number of objects bounding boxes with the significant achievements made in object tracking.

D. Human Motion Tracking

The main work of motion tracking is to find detected object in each frame and search motion path of the object to track in time sequences. That is changing location with respect to its background. By using inference algorithm. To increase reliability and robustness, analyzing temporal information is good track for trajectories. The motion sequences move each target from frame to frame, which is the key of moving object tracking to locate the search regions of a target in the next frame. When the next frame, this distribution has changed due to the new mode and tracks the object correctly. This system gives a high precision of detection and tracking for moving object concerned with variation of appearance such as presence of noise in foreground image, different poses, changes of size, shape and scene in indoor or outdoor environment. Input video and result of outdoor environment as shown in Fig. 4.

IV. Human Motion Recognition

In human recognition systems, the input features have structured in various ways in motion

sequences. Since human actions are highly dynamic, partially visible, occluded, or cropped targets and very closely resemble with ambiguities. For the understanding and analysis of human behavior, recognition with skeleton system are robust to variations of viewpoints, body scales and motion speeds. It is essential for general shape representation and that information can be access in real-time. The system first extracts simple but effective frame-level features from the skeletal data and build a recognition system based on deep learning neural network to obtain the recognition results on defined action sequence level. Human body parts representation from spatial-temporal parts as expressed in Fig. 3.

A. Deep Learning Approach

The system applies Deep Neural Network (DNN) to represent human in 3D space and to recognize skeleton joint locations. DNN is capable of capturing the full context of each body joint in the full image as a signal. Since, skeleton feature points are scattered in human body and joint coordinates system has different coordinates due to translation and rotation and then depend on different body sizes. The system is learned in neural network for connecting the skeleton feature points and estimating joints to increase precision result of joint localization.

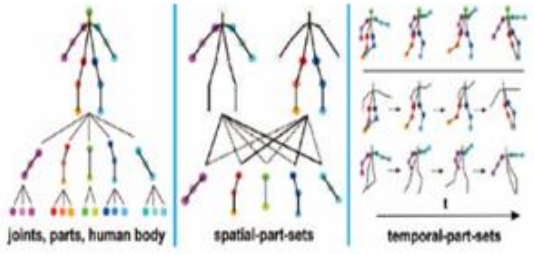


Figure 3. Human body parts representation from spatial-temporal parts

B. The Gradient of Distance Transform

Human body is connected by joints and human action present continuous evolution of spatial configurations of joints [9]. We normalize the positions of the joints of each subject using the neck point and then concatenate (x, y) coordinates into a feature vector. The system is trained in a deep network using similar approach to the above shape context method. To obtain the connected lines, we may need to explore the local features that is using with the gradient of distance transform image, as in:

$$\nabla DT=\left(\frac{\partial DT}{\partial x},\frac{\partial DT}{\partial y}\right) \quad (5)$$

The norm of gradient vector is obtained in:

$$|\nabla DT|=\sqrt{\left(\frac{\partial DT}{\partial x}\right)^2+\left(\frac{\partial DT}{\partial y}\right)^2} \quad (6)$$

To overcome unmatched point in skeletonization, the system is defined on constraint condition when searching the successive feature points that means $DT(p)>0$, where p is a rigid part that is joined by joint. It is reliable for any lines connecting two points must locate on the foreground. Roughly, the proportions of human body parts are the same.

C. Skeletonization

Evolving the skeleton feature points in the body parts is a specific routine of skeletonization. By capturing spatial-temporal representation of body parts in one frame as well as movements across a sequence of frames through graph model [10]. The graph model captures the geometric and appearance variations of human at each frame and that represent the motion of human with 3D skeleton joints. The structural data is represented by pairwise features, relative to the position of each pair of joints relate to each other. The orientation between two joints i and j are computed, as in:

$$\theta(i,j)=\arcsin\left(\frac{i_x-j_x}{dist(i,j)}\right)/2\pi \quad (7)$$

where $dist(i,j)$ is define for the geometry distance between two joints i and j in 3D space.

Figure 4. Input walking video and recognition result in outdoor environment

V. Experimental Results

The results obtained in the implementation are shown in this section by using HMDB51 dataset consists of 6,766 videos. That is collected from various sources from YouTube, Google videos, mostly movies and small public database. From that we experiment on 500 videos with 4 actions. The frame rate of videos is 30 fps. Testing on dataset with different conditions are shown in Table I. The confusion matrix on different activities are described in Fig. 5. The performance of the system is critical point in any dataset. The accuracy for human recognition results on different iterations are expressed in Fig. 6.

VI. Conclusion and Future Work

In many application area, real-time information is very important and require for efficient human motion detection, tracking and activity recognition system. That can record useful information and analyze the environment in many scopes. There are many challenges that are concerned with different variation in human pose, shape, illumination changes and background appearance. In this paper, the system is implemented by using deep neural network framework to get high accuracy recognition of human movement in indoor and outdoor areas. The experimental results have been concluded that all method have a big dependency on different backgrounds, camera calibration and illumination changes. We trained and tested video data on different changes that are significantly increased the detection, tracking and recognition rate of our results.

Future research directions will continue 3D skeletal model for moving object with various dataset containing different activities to describe the accurate result of human motion detection, tracking and activity recognition system.

TABLE I. Testing The Sequences on HMDB51 Dataset

Activity	Frame Per Second	Length of Time	Number of Track Box	Conditions
Walking	30	00:00:05	12	sunny
Standing	30	00:00:03	13	indoor
Sitting	30	00:00:05	10	cloudy
Running	30	00:00:10	5	outdoor

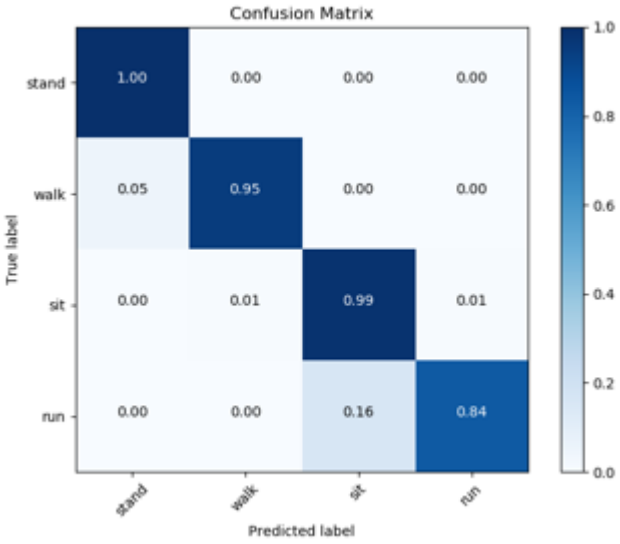
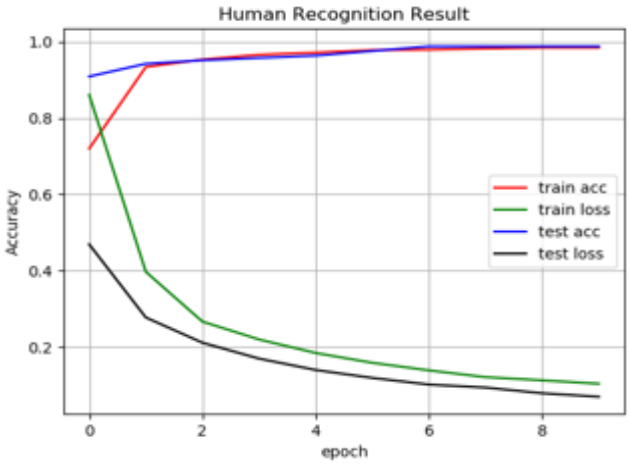


Figure 5. The result of detection, tracking and recognition with confusion matrix on different activities



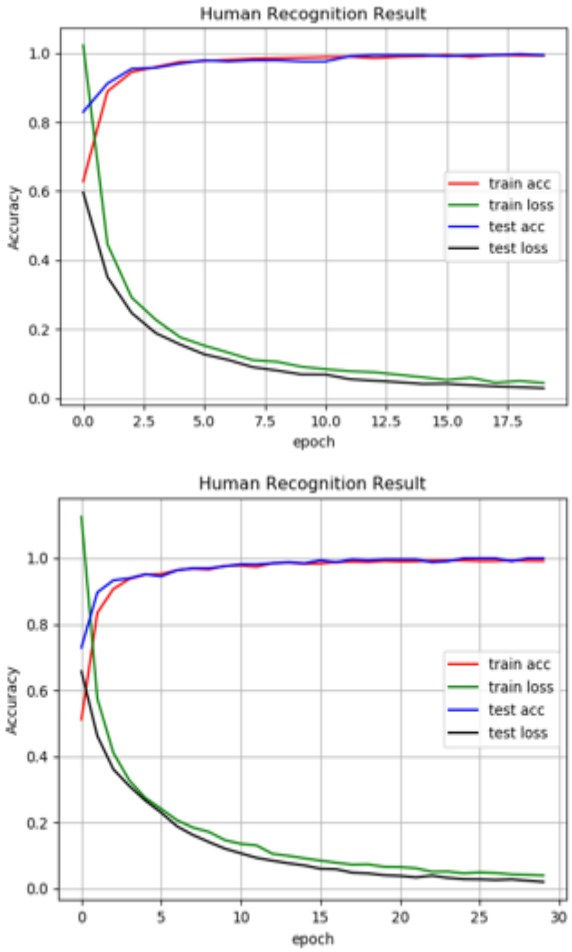


Figure 6. Human recognition result on training and testing with different iteration

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Vehicle Accident Detection on Highway and Communication to the Closest Rescue Service

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Abstract

The hazard information and the timely rescue performance are considered as the main elements to reduce the risk of road traffic accidents since the rate of highway accidents significantly increased. In this paper, it is aiming to detect the highway accident victims by using the data received from Sensor Fusion-Based algorithm while Ray Casting algorithm will assist the users to receive the assistance of rescue services in timely manner. With the purpose of user friendliness, these algorithms are intended to apply in the smartphones built-in high technology sensors, which are connected with the GIS, GPS and Geofence technologies.

Keywords**:** *Highway, Accident Detection, sensors, Rescue Alert, GPS, GIS, Geofence, Dataset*

I. INTRODUCTION

In Myanmar, road collisions are the major cause of death among the young adults and mid-aged people. Most of the precious lives are lost during the road accidents because it is not able to identify the correct location of accident occurred as well as inform the rescue services to save the victims on time. A highway is the main road in which connects the urban and rural areas. Due to its wide structure and possesses the high-speed thresholds, it can reduce the travelling time of the users. In Myanmar, there are forty- nine (49) roads, which connect from Eastern Regions to Western Areas, while the total of thirty-six (36) roads are used as highway to link from Northern States to Southern Provinces. The total length of these roads is more than twenty thousand (20,000) kilometers. All of the matters related to the usage and maintenance of these roads are monitored and managed by the Department of Highways, which is under the Ministry of Construction in Myanmar. Recently, it is noticed that the rate of fatality and injury on Highways in Myanmar has been an upsurge. According to the statistical results stated in "Road Safety in Myanmar 2017" by Federation Internationale de l'Automobile (FIA), the number of deaths related to road accidents received from Myanmar Police Force was 1,853 in 2008 and, this number was increased to 4,688 in 2016. Within eight years, the numbers of reported road deaths were doubled.

Even though the economy and development of Myanmar is still in growth, the vehicle ownerships of the citizens are increasing all the time. It is expected that the number of accidents related to automobiles are likely to be raised and, is estimated that 9,000 people may be killed in road accidents in 2020. Apparently, there is a linkage between a person's death during the above accidents and the absence of the first aid provision due to the delay in informing to the rescue services. Thus, in the case of road accidents, response time is vital for saving the precious lives of accident victims and considered the certain level of impact on death rate. In Myanmar, the smartphones embedded with strong detection sensors to identify the information of accidents and send this to the responsible teams in timely manner are not available yet.

With the intention of overcoming the weaknesses in the Rescue Alert System via Geofence technology, this research aims to validate the status of accident occurred by using the values received from accelerometers and gyroscope sensors, identify the accurate location of accident by using GIS and GPS technologies, maintain and update the rescue service information such as contact points, and search for the nearest rescue units with the usage of Geofence Technology. In this way, the fatality rate due to the vehicle accidents will be reduced to the certain level as the prompt response is received from the emergency services such as Medical emergency services. Once the location of the accident is confirmed, the Ray Casting Algorithm is applied to search for the closest rescue services within the shortest time duration. The similar experiments related to the vehicle accident detection

and communication to the rescue services are also as followed:

Chris T., White J., Dougherty B., Albright A. and Schmidt DC.[1] have developed a prototype smartphone- based client/server application "WreckWatch", which operates together with the embedded smartphone sensors and communication interfaces to detect the causes of accidents and deliver the notifications to the respective organizations. The vulnerability of this application is the speed limitation and, the smartphone application does not detect the possibility of accident in low speed condition when the speed of the car is lower than the configured speed threshold in application. Sneha R.S. and Gawande A.D [2] invented the accident notification system called "E-call" for smart phones. This "E-call" uses the cellular network to transfer the data between smartphones and the Server Center. Since "E-call" system accessed the built-in accelerometer sensor of smartphone as an accident detection sensor, it is possible to produce the false positive alarms although the user is not inside the car. In 2015, Dipesh Suwal, Suresh Manandhar, Shashish Maharjan and Ganesh Dhakal, [3] "D-Fencing Application", which notified the Geo-fencing Post Disaster Scenario using Android App. The users will receive the alert messages from the system admin whenever they are approaching or entering to the regions affected by disaster.

However, the information in database may not be updated because only the system administrator, a person who has knowledge of Geofence, can manage the disaster information as well as send the alert message to the requestors. If the system administrator is not available when the incident occurs, the users may not receive the updated information. Hence, the application is not reliable since it does not have the automated features of receiving disaster information, updating the database promptly and sending the precise rescue alerts to the users.

In 2013, Danish karim and Jaspal Singh [4] proposed the "Development of Automatic Geo-Fencing and Accidental Monitoring System based on GPS Technology". By operating a single shock sensor embedded in a proposed system, an accident can be detected as well as the vehicle can also be prevented from the theft as it is operated as security control. However, there is no predefined database and datasets to record the user information and the variables of incident occurred. Although it was mentioned that the system should send the alert automatically to the rescue services, it was not mentioned how and where the contact information of rescue services is maintained. Thus, the recipients were likely to receive the limited information of sender via SMS. The remainder of the paper is organized as follows.

In section 2, explain the methodology of this research study. In section 3, we discuss the incident prediction of sensor and driver prediction by using sensor fusion-based algorithm used together Ray Casting algorithm and the elements of Geofencing Technology. In section 4, explain the technical usage of this proposed system. In section 5, the experiments and results are explained. Finally, section 6 presents the conclusion.

II. METHODOLOGY OF RESEARCH STUDY

The performance of an accident detection system is determined levels: data collection with data processing and communication to the closest rescue service is illustrated in Figure 1. Firstly, the values received from Accelerometer, Gyroscope and GPS are maintained in the accident detection dataset. Secondly, the system predicts as an accident and the accident validation message will be triggered to the user when these values are exceeding the accident threshold value. Thirdly, the system starts to search for the accident location by using the GPS values received from the accident detection dataset when the accident is confirmed. Fourthly, the Geofence technology creates the polygon in which the accident location is marked as the center point. Fifthly, the system accesses the rescue service dataset to retrieve the contacts of rescue services located in the polygon and, search for the closest rescue service from the accident location.

The Geofence will create another polygon with wider boundaries if none of the rescue service contacts is identified in the first polygon. Finally, the system will send the alert message including the user and vehicle information as well as the location of the accident to the rescue service if it finds the closest one from the accident location.

The server side's workflows are emphasized to collect and process the data including the users' information, incident information and the contact of rescue services. And then decide the relevant action while user side workflows act as the projection of the prediction. To make more accurate prediction, sensor projected data such as false alarm rate; fatalities of accidents, fixed data are stored in centralized system and communication differences parties, smartphones are play as sensor roles to make prediction.

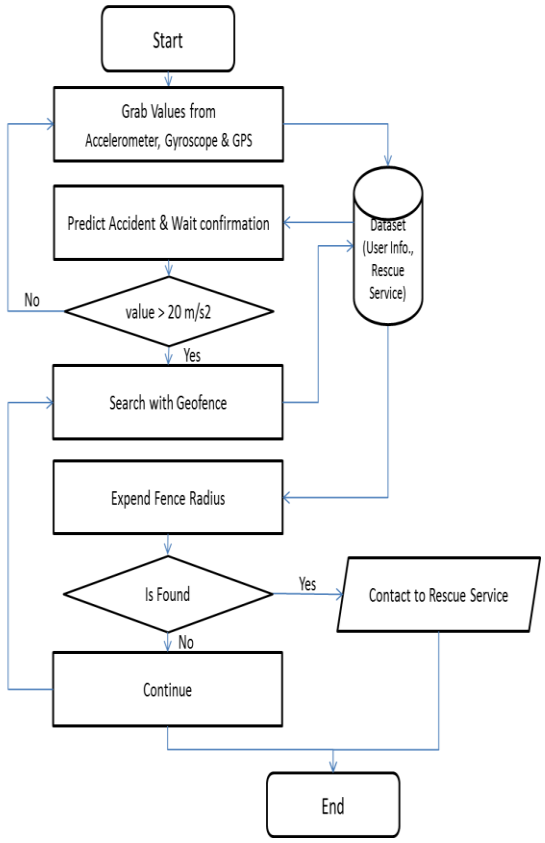

Figure 1. System Architecture

III. DETECT AND SEARCH ALGORITHMS

A. Sensor Fusion - Based Algorithms

Since the data deficiency issue occurs among the individual sensors, Sensor Fusion-Based algorithm is considered as a compensating method to enhance the data reliability by merging the data from various sensors. According to the previous similar experiments, the information produced from the automated sensors might be unreliable to some extent. In Sensor Fusion-Based algorithm, the information can be retrieved not only from automated sensors but also from historical sensor data stored in the centralized database as well as the non-automated sources. Then, the data received from multiple sensors will be refined and evaluated to achieve the optimal result. In this accident detection and rescue alert system, it is crucial to obtain the high-quality input data.

The most favorable result to detect an accident can be obtained by using the following Sensor Fusion - Based algorithm:

$$Xi\ (t) = Ai\ xi\ (t) + Bi\ ui\ (t) + Wi\ (t),$$
$$yi\ (t) = Ci\ Xi\ (t) - vi\ (t),\ (i = 1, 2, n)$$

Where i indicates the number of subsystems (i.e. accelerometers, gyroscopes and so on) in which data is produced and t indicates the time period.

The present value of the subsystem Xi(t) can be expressed by combing the current condition of the source xi(t), the signal value transmitted at the particular time ui(t) and, the noise value at the time period Wi(t). When the value of Xi(t) is received, the output of the subsystem or source yi(t) can be determined by filtering the unwanted noise value vi(t) included in present value of Xi(t).

By using an MLF-type neural network, the most favorable result to detect the accident will be predicted upon the individual output values as well as the integrated result of those individual output values.

B. Ray Casting Algorithms

When the status of accident is confirmed, a polygon fence with the range of 1,000 square meters will be created and the accident location is marked as the center point in that range. In this fence, the contacts of rescue services near the accident location will be searched with Ray Casting Algorithm. If the rescue services cannot be found, the Geofence will re-create the new polygon fence with wider boundaries and search for the contact points of rescue services with Ray Casting Algorithm until the closest rescue service is identified. When the closest rescue service is found, the accident alert message including user and vehicle information as well as the location of the accident will be triggered to the closest rescue services. The intention of applying Ray Casting Algorithm in this paper is that the closest rescue services from the accident location can be found and contacted with a minimum time interval.

In the Ray Casting algorithm, there are three input values, P, Q and R. P defines the point in which the place of accident occurred while Q states the polygon created to encircle P. R indicates the locations of rescue services marked in Google Map. Although the initial value of inside is set as ‘False’ at the beginning of every search, the final value of inside will be shown as ‘True’ if R falls in Q and, ‘False’ if R is out of the edges of Q.

Initially, the system sets the count as ‘0’ before starting the looping of every edge from point P. S indicates the ray that is originated at point P and transmitted to +y direction and, this ray will be only in the edges of y direction. When the accident is detected, the infinite rays or S will be transmitted in +y direction from its point of origin or P. To verify the target point of interest or nearest rescue service (R), there will be

a looping between P and the edges of polygon, Q. Once the point of interest (R) is identified in Q, the value is marked as 'True' and then, the searching process is ended. If the point of interest (R) is not fallen inside the Q, the value is marked as 'False' and the looping will be stopped. Then, another polygon with larger diameter will be created to start the looping until the nearest rescue service (R) is found.
Algorithm:

Input: Points as P, Polygon as Q, Rescue Service as R
P is the position of interest
buf is a buffer distance.
Output: true if R contains Q, otherwise false
1: count = 0
2: s is an infinite ray in the +y direction, originating at P
3: for all edges e in Q do
4: if R is within buf of ex then
5: ex,buf = ex -2 * buf
6: else if R is within buf of e or ebuf then
7: return false

C. Geofence

Geofence creates the virtual geographic boundary of an area where the accident occurs. To mark a location of interest, the user needs not only to specify its latitude and longitude but also add the adjustment of radius. Once the latitude, longitude, and radius are provided, it is defined a geofence (a circular area or fence) around the location of interest. This feature is to use for hazard area detection. In this paper, geofence is applied to send the information to the closest rescue team from the geographical location where the accident occurs.

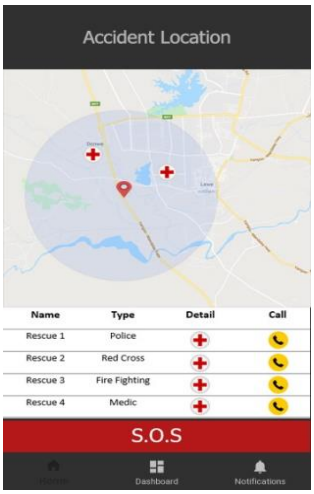

Figure 2. Information of Rescue Services Location and Accident place

D. User Information

Users are required to input information to process for faster and, receive efficient rescue procedure. Before starting the trip on highway, the users including driver and passengers must provide their personal information such as name, gender, blood type, date of birth, contact number and emergency contact number. This information is collected upon installing the one-time input. For example, when fatal accident occurs, the victim might need blood, so the rescue team can prepare for the required blood type of victim.

E. Vehicle Information

After providing the personal information of the users, it is also required to insert the information of the vehicle that the user drives on the highway. Once the accident occurs, the information such as the accurate position is provided upon the selected route of the trip when the user activates the accident detection system. Then, the system will be able to access the below data before informing the rescue teams.

- Car Type (Sedan, SUV, Truck, Wagon, etc.)
- Vehicle Registration Number
- Color of Vehicle
- Number of Passengers

F. Accident Detection Dataset

These datasets are early stages as the built-in sensors of the system need to retrieve and evaluate the status of the accident. Once the status is confirmed, the latest calibration sensors data is delivered to the main system to create the updated dataset automatically. The accident detection datasets decide upon the values received from the smartphones' built-in sensors such as the accelerometer, gyroscope and linear acceleration whether the accident may occur or not.

G. Rescue Service Dataset

With the connection of Geofence, Rescue Service Dataset maintains the data about the rescue services such as the contacts of ambulance service, fire stations, and police stations and so on. In figure 3, the sample of rescue datasets are illustrated. Once the accident is confirmed, then the nearby rescue services will be informed after automatically adding the details of last GPS. These datasets are constantly updated in the servers, thus Geofence can access whenever it is required.

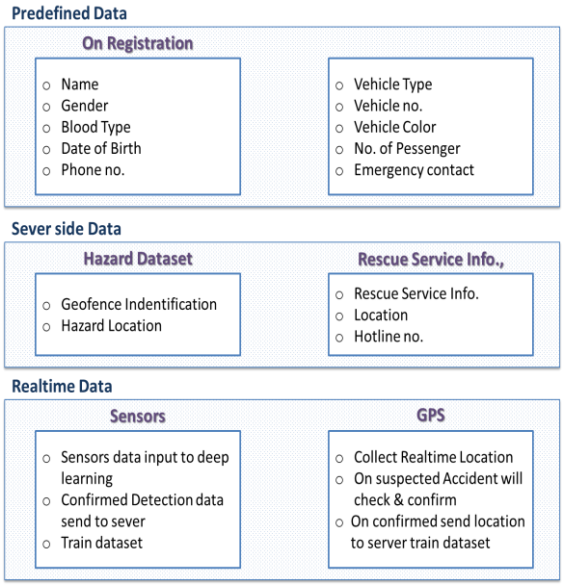

Figure 3. Sample of Sever side Dataset

IV. TECHNOLOGIES USAGE

The following technologies are intended to use for detecting the accidents and sending the alert to rescue teams.

Smartphones are chosen as the major subject of this research as they are the affordable and effective portable machines embedding with powerful technologies. Being an open-source mobile operating system, the Android source code is free to access with only one restriction, which is not allowed to use for personal interest or any other financial benefits. Being the primary Android IDE (Integrated Development Environment), Android Studio supports an Android developer all the required tools when the Android applications are developed. Hence, the developers can write the codes with auto-completion tools.

It is more productive for developers to use Java than any other languages as Java has the large set of class libraries that comprise the multiple functions such as connectivity, sensors and so on. These libraries are reusable by most of the embedded applications, so the developers do not need to rewrite the codes, and this could not only save the developing time but also prevent the unexpected errors.

Geo-fencing creates the virtual boundaries for the actual geographic locations. The virtual boundaries could be fixed or changeable. In a comparison with the conventional framework API, Google Location API delivers the more powerful, high-level framework that can select a suitable location provider and power management without the assistance of manpower. One of the prominent features of this API is the activity detection which is not available in the framework API.

The Google Maps Android API is opted for this research as it is a service allowed to access the Google Maps Server automatically to present Maps, record the response in the target location and retrieve the required information for the user. With the usage of HTTP request, this service provides the information about the geographical locations and, adds the noticeable points of interest along the driving route. Unlike Google Places API Web Service, Google Directions API calculates the distance between the starting point and the destination and, presents the most efficient routes based on travel time, number of turns, distance, etc.

Elements in Geo-fencing

GPS: It can pinpoint the location accurateness within 5 meters from the user's location. Besides, GPS is accessible to all users without burdening on carrier infrastructure.

A-GPS: Assisted GPS (A-GPS) supports to obtain the most accurate result of mobile positon by evaluating the GPS satellite information with the data received from mobile tower sites.

Cell-ID (COO/CID): Mobile device can be easily detected by pinpointing the connected cellular network tower. Since, Cell-ID is uncomplicated way, and it is not necessary to improve network location infrastructure and mobile phones.

Wi-Fi: Wi-Fi is suitable for both indoor and outdoor environments as it decreases time-to-first-fix (TTFF) as well as is highly accessible in urban regions.

V.EXPERIMENTS AND RESULT

The lack of treatment in the proper time is the major reason for many deaths. The major causes may be the late arrival of ambulance or no person at the place of accident to give information to the ambulance or family members. This paper is intended to reduce these factors and save the lives of victims by delivering the required medical assistances to accident location on time.

The possibility of accident can be predicted immediately based on the values received from the accelerometers and gyroscopes by accident detection system. If these values exceed the predetermined accident threshold in the system, the validation message will be sent to the users whether accident is occurred or not. In general, the system will wait for maximum twenty (20) seconds to receive the confirmation from the user. Whether the confirmation from the user is received or not within twenty (20) seconds, the system starts to create the visualized

polygon near the accident spot by using Geo-Fencing technology. With the visualization facility of Geo-Fencing Technology, not only the accident spots can be discovered for timely response but also can be assisted to verify the nearest locations for the rescue service facilities. Then, the system access to rescue service dataset and retrieves all the contact spots of rescue units within the Polygon.

By using the Sensor Fusion-Based Algorithm and the Ray Casting Algorithm, received values from the sensors are accident can be predicted and then the closest rescue service spot from the accident location is searched with minimum time interval as it is crucial to deliver the rescue team to the accident victim in timely manner. Once the closest rescue contact is spotted, the system sends the alert message including the list of accidents happened around that selected rescue service location.

To improve the performance of the sensors and obtain the accurate results, the phone must be docked in the vehicle. It is essential to record the movement values, which is standardized as 5 g, on a timely manner. In general, the accident will be identified when the value of X axis becomes greater or smaller than the values of Y axis and Z axis within 0.6 second. It can be deemed as a fatal accident if the value of X is significantly greater than the two other values, Y and Z, after one second.

In Figure 4, the 3-Axis accelerometer is applied to measure the accelerations of the sensors while the car is out of control and is rolling. In this case, the acceleration value of X-axis exceeds the range of -5g and +5g.

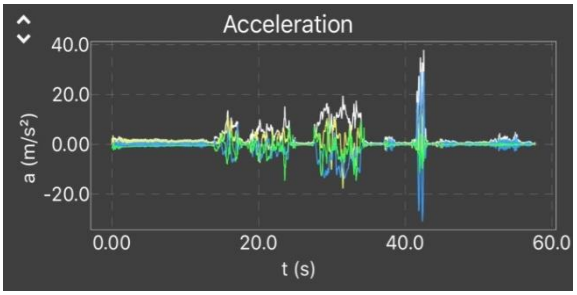

Figure 4. The car rolls as it exceeds the speed threshold and out of control

Moreover, it is also required to maintain the unique client-server architecture to collect and retain the updated data regarding about the real-time situations occurred on highway. Thus, the users from different sectors can access and verify the information effectively and efficiently. This system is planned to implement as a user-friendly version for the users to share their information before starting the trips.

In figure 5(a) and 5(b), the users require registering about user's personal data as well as the general information of the vehicle using for this trip. The requested data indicated in these figures are needed because the required procedures can be prepared such as the correct blood types for the accident victim.

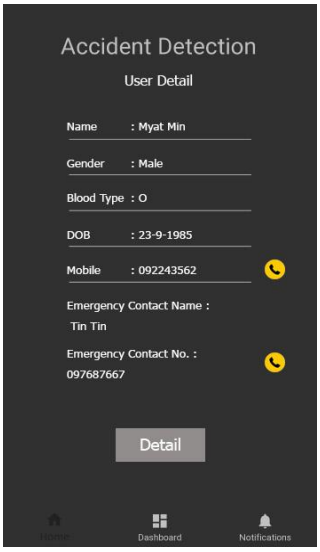

Figure 5(a). Registration of Personal Data

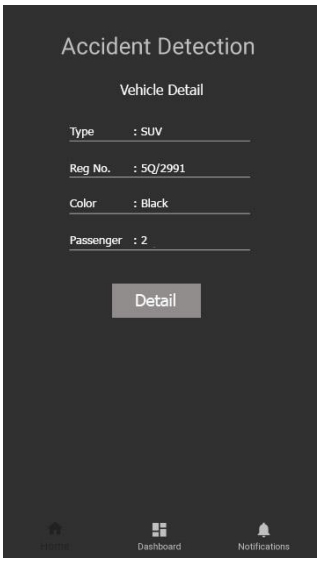

Figure 5(b). Registration of the desired trip

All information of users and vehicles will be maintained in the Database, and the system will obtain them whenever it is required. In figure 5(c), the rescue

service, which is the closest to the accident location, will receive the alert message from the system. In this message, the data of user including emergency contact as well as the notification of other accidents received from database will be observed. The latest accident will be shown on top of the list.

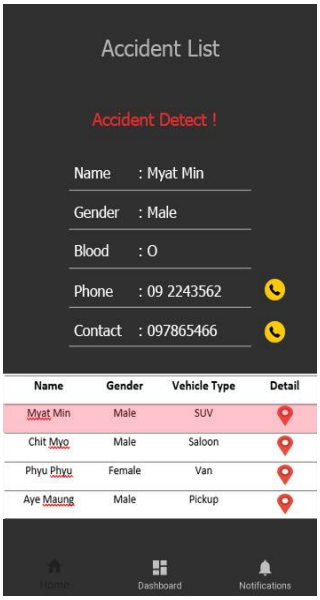

Figure 5(c). Accident Information

VI. CONCLUSION

When the rescue procedures presently used are verified, it is noticed that the current accident detection solutions including smartphone systems still have some weaknesses such as the speed threshold limitation, the higher false alarm rate, the manual controls upon updating the database information, and the lack of predefined dataset to analyze the accident data.

In this paper, the proposed system will be compatible with the various trained dataset as well as operating with Google application programming interfaces (APIs), and thus, the weaknesses mentioned above will be solved. Once the well-trained datasets are maintained in the systems, the user's conditions while using the highways are tracked and updated in the database perpetually. Thus, the system can obtain those data to validate the status of accident whenever it is needed. Moreover, with the usage of Geo-fencing and the Google APIs, the location of accident can be accurately identified within a short time and, the closest rescue services from the accident spot will be pinpointed and informed to save the victims of accidents.

Since this system is implemented as the user-friendly version, the user can simply reply by pressing the verification message whether he or she needs the assistance of rescue team or not. Even the user is unconscious; the system will wait for certain time duration before contacting to the nearest rescue unit via Geo-Fencing Technology to prepare the required procedures. Moreover, the system always synchronizes with user's device and even the device is switched off, the last information of users is still captured in the system to verify the data of accident.

ACKNOWLEDGMENT

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Natural Language and Speech Processing

A Study on a Joint Deep Learning Model for Myanmar Text Classification

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Abstract

Text classification is one of the most critical areas of research in the field of natural language processing (NLP). Recently, most of the NLP tasks achieve remarkable performance by using deep learning models. Generally, deep learning models require a huge amount of data to be utilized. This paper uses pre-trained word vectors to handle the resource-demanding problem and studies the effectiveness of a joint Convolutional Neural Network and Long Short Term Memory (CNN-LSTM) for Myanmar text classification. The comparative analysis is performed on the baseline Convolutional Neural Networks (CNN), Recurrent Neural Networks (RNN) and their combined model CNN-RNN.

Keywords—*text classification, CNN, RNN, CNN-RNN, CNN-LSTM, deep learning model.*

I. INTRODUCTION

Text classification is one of the interesting application areas in NLP such as sentiment analysis, machine translation, automatic summarization, etc. Nowadays, information overloading is one of the important problems for today's people. People waste too much time to select their interesting information because they receive a vast amount of information from various sources of internet. A powerful text classifier can extract useful content from massive amounts of information. It can classify text into associated categories including sentiment analysis, spam detection, articles, books, and hate speech detection. It is among the most active research areas in the field of NLP. Historically, many researchers performed text classification by using various machine learning algorithm and their variant models. The use of deep learning models has achieved great interest in text classification due to their ability to capture semantic word relationships in recent years. All deep learning models typically require a lot of data to accomplish a specific task. Although most pre-trained vectors are trained on general datasets, the use of pre-trained vectors can handle data requirements. It can be used for the specific task by transferring learning with the appropriate amount of data. This paper focuses in particular on the classification of Myanmar articles. Words are considered as basic units in this paper. Since Myanmar language has no standard rule to determine word boundaries, it is important in the pre-processing phase to determine the word boundaries. As Myanmar language is a morphologically rich language, good word representation is difficult to learn because many word forms seldom occur in the training corpus. The BPE tokenizer[1] is used to determine the boundary of the word. The segmented words are converted into an embedding matrix by using the pre-trained vectors trained on Wikipedia by the Skip-gram model [2]. The effective use of pre-trained vectors is very supportive of resource-scarce languages. In this paper, CNN and LSTM models are jointly performed for Myanmar text classification. The convolution process is performed on the embedding matrix for selecting the features and the LSTM model is used to capture long term information. In this paper, the max-pooling layer is dropped in the CNN model to prevent the loss of context information.

The next sections are as follows, the works related to the text classification and other areas of application are addressed for both Myanmar and English in section 2. Section 3 describes Myanmar's background knowledge. Section 4 discusses the components of the CNN-LSTM joint model. Section 5 explains the findings of the experiment and the paper is concluded in section 6.

II. RELATED WORK

Wang et al. [8] proposed a regional CNN-LSTM model that captures regional information by CNN and predicts valence arousal (VA) by LSTM

[1] https://github.com/bheinzerling/bpemb

model and outperforms than regression-based, lexicon-based and NN-based methods on two datasets for sentiment analysis. Kwaik et al. [5] proposed a deep learning model that combines LSTM and CNN models for dialectal Arabic sentiment analysis and this model performs better than two baseline models on three datasets. Zhang et al. [9] constructed the CNN model on the top of LSTM. The output from the LSTM model is further extracted by the CNN model and it performed better in terms of accuracy than the baseline models. Kim et al. [4] conducted the experiments with variations of CNN model, CNN-rand, CNN-static and CNN-multichannel on the top of pre-trained vectors for sentence classification. These CNN models performed better in 4 out of 7 tasks than the state-of-the-art.

The previous research works in deep learning and machine learning models for Myanmar language in speech recognition, named entity recognition, and text classification are also investigated. Aye et al. [1] aimed to enhance the sentiment analysis for Myanmar language in the food and restaurant domain by considering intensifier and objective words and improved the prediction accuracy. Hlaing et al. [3] applied LSTM-RNN in Myanmar speech synthesis. Mo et al. [6] annotated Named Entity tagged corpus for Myanmar language and evaluation was performed comparatively between neural sequence model and baseline CRF. In our previous works [7], we performed the comparative analysis of CNN and RNN both on syllable and word level by using three pre-trained vectors and also collected and annotate six Myanmar articles datasets. In this paper, a comparative analysis is performed on a joint CNN-LSTM model against baseline CNN, RNN, and their combine model by using Myanmar articles datasets.

III. MYANMAR LANGUAGE BACKGROUND

Myanmar language is the official language of the Republic of the Union of Myanmar. It is a morphologically rich language and Myanmar sentences are basically constructed as the subject, object, and verb pattern. The Myanmar script is written from left to right and the characters are rounded in appearance. There is no regular inter-word spacing in Myanmar language like in the English language. Though, spaces are used to mark phrases. Sentences are clearly delimited by a sentence boundary marker. Myanmar words are constructed by one or more syllables. There are thirteen three basic consonants, eight vowels and eleven medial. A Myanmar syllable is constructed by one initial consonant (C), zero or more medial (M), zero or more vowels (V) and optional dependent various signs. Words are considered a basic unit in this paper.

A. Pre-processing

Pre-processing basically contains two steps, 1) removing unnecessary characters, and 2) determining the word boundaries. As this study focus on the Myanmar text classification, Myanmar Unicode range between [U1000-U104F][2] is removed to ignore non-Myanmar characters. The numbers "၀-၉", [U1040-U1049] and punctuation marks "၊, ။", [U104A- U104B] are also removed. As Myanmar language has no rule to determine word boundary, it is needed to determine the boundary of words. In this work, word boundaries are determined by the BPE tokenizer.

B. Pre-trained Model

The reuse of the pre-trained model by transfer learning on a new task is very efficient because it can train the deep learning models with not much data. Typically, most of the NLP tasks don't have sufficient label data to be trained on such complex models. In this paper, we use the pre-trained vector file that was publicly released by the Facebook AI Research (FAIR) lab[3]. It was trained on Wikipedia using the fastText skip-gram model with 300 dimension. In this paper, the pre-trained model is used as the starting point instead of learning from scratch.

C. Construction of Embedding Matrix

The segmented words are transformed into word vectors by matching the vocabulary in the pre-trained vectors file. Figure 1 shows the construction of the embedding matrix. Firstly, Myanmar text data are extracted from online news websites. Secondly, unnecessary characters are removed from the extracted text. Then, word boundaries are determined by the BPE tokenizer although sometimes the results are meaningless as it is a frequency-based tokenizer. The tokenized words are matched with pre-trained word vectors file in order to construct the word embedding matrix for the embedding layer.

Table I shows the sample of Myanmar text pre-processing with the sample sentence "Corona ဗိုင်းရပ်စ်

[2] https://mcf.org.mm/myanmar-unicode/ [3] https://fasttext.cc/docs/en/pretrained-vectors.html

ကာကွယ်ရေး မြန်မာတိုးမြှင့်ပြင်ဆင်။” and non-Myanmar characters “Corona” is removed. Then, the text string is segmented into words as “ဗိုင်းရပ်စ်_ကာကွယ်ရေး_မြန်မာ_တိုးမြှင့်_ပြင်ဆင်”, “_” shows the boundary of words.

TABLE I. SAMPLE OF TEXT PRE-PROCESSING

Input Sentence	Corona ဗိုင်းရပ်စ် ကာကွယ်ရေး မြန်မာတိုးမြှင့်ပြင်ဆင်။
English Meaning	Myanmar ramps up the defense against Coronavirus
Word Segmentation	ဗိုင်းရပ်စ်_ကာကွယ်ရေး_မြန်မာ_တိုးမြှင့်_ ပြင်ဆင်

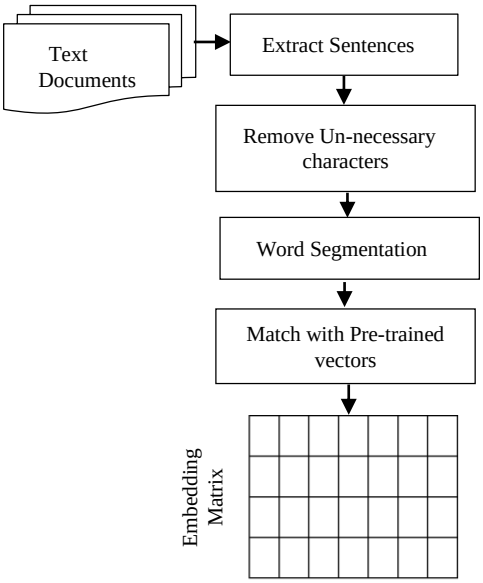

Figure 1. Construction of embedding matrix.

IV. A JOINT CNN-LSTM MODEL

The proposed joint CNN-LSTM model basically consists of four components, 1) Embedding Layer, 2) Convolution Layer, 3) LSTM layer, and 4) Fully connected and output layer. Figure 2 illustrates the joint CNN-LSTM model.

A. Embedding Layer

In the embedding layer, segmented words are transformed into vector representation by matching the pre-trained vectors file. The pre-trained vectors file is like vocabulary and each word in the vocabulary attached with their corresponding vectors that can catch the context information.

B. Convolution Layer

In the convolution layer, the convolution process is performed by the ReLU activation function with stride size 1. Convolution layer selects the features and the result of the convolution process is in the form of the feature map. Max-pooling layer is discarded because it captures only very important features and ignores the un-important features and it can lead to losing the context information.

C. LSTM Layer

LSTM layer is used instead of the max-pooling layer to capture context information.

D. Fully Connected Layer and Output Layer

In the final output layer, the probability of the class is predicted by using the sigmoid activation function. In the hyper-parameter setting, Adam optimization function and binary_crossentropy loss function with 0.5 dropouts and 16 batch size on 10 epochs. Moreover, l2=0.01 is set in kernel and bias regularizer to reduce overfitting.

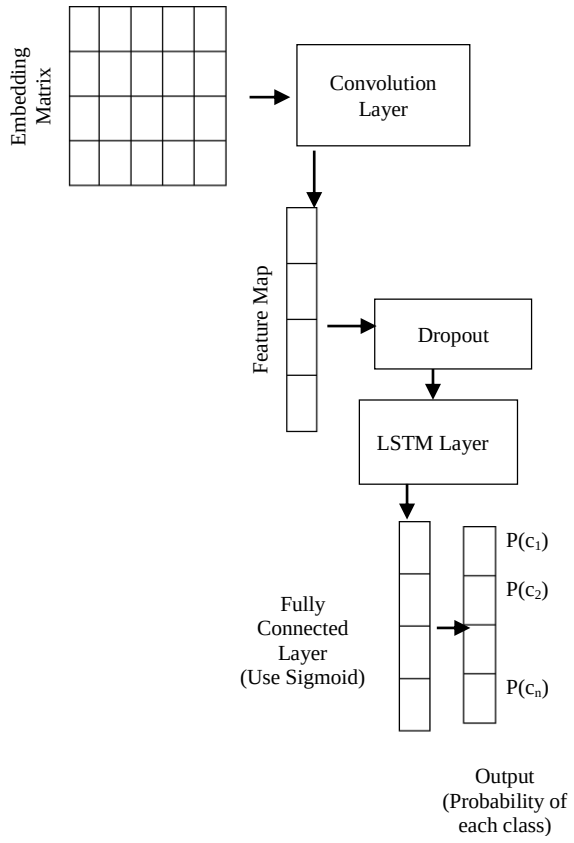

Figure 2. A joint CNN-LSTM model

V. EXPERIMENT

A. Dataset

Myanmar text data are collected from various sources of Myanmar news websites including 7Day Daily[4], DVB[5], The Voice[6], Thit Htoo Lwin[7], and Myanmar Wikipedia[8]. The text data contains five categories art, sport, crime, education, and health. Each category is collected in tab-separated values (.tsv) files and each row contains a sentence that is annotated with the corresponding label. The text data are converted from Zawgyi to Unicode by Rabbit converter[9] and shuffle and split 75% and 25% for train and test dataset for each topic and the total number of sentences for the training is 36,113 sentences and the testing is 12,037 sentences. The details of the dataset are listed in Table II.

TABLE II. MYANMAR TEXT DATASET FOR FIVE TOPICS

Class	Train	Test	Total
Sport	9,919	3,242	13,161
Art	9,483	3,173	12,656
Crime	6,718	2,184	8,902
Health	4,743	1,632	6,375
Education	5,250	1,806	7,056
Total	**36,113**	**12,037**	**48,150**

B. Comparison Models

In this work, the comparative analysis is performed on a joint CNN-LSTM model with baseline CNN, RNN, and their combined model CNN-RNN.

1) CNN: CNN is a feed-forward neural network and a basic CNN contains three layers, convolution layer, pooling layer, and fully connected layer. The ReLU activation is mostly used in the convolution layer, max-pooling is mostly used in the pooling layer and Softmax function is mostly used in the fully connected layer. In this paper, the simple CNN with one convolution layer and sigmoid function is used in the experiment.

2) RNN: RNN is an artificial neural network with an internal memory that keeps information to persist. It learns from the previous data and performs the same function for all input data. RNN produces the output y_t as in equation[10] (1).

$$y_t = f(W_y h_t) \quad (1)$$

$$h_t = \sigma(W_h h_{t-1} + W_x x_t) \quad (2)$$

3) CNN-RNN: In this combined model, the CNN model is used for feature extraction and RNN is used to make a prediction by past words. The drawbacks CNN model is its locality and it requires many convolution layers to capture long term information. RNN is a bias model and it predicts the semantic of words by past words and reduces performance when predicting long text. LSTM model is an extension of the RNN model to be better preservation of long term dependency problem.

C. Experimental Setup

The experiment is implemented on Google Colaboratory (Colab) [11] that provides the Jupyter notebook environment[12], executes code in Python 3.6.9. No setup is required and runs in the cloud and have a maximum lifetime for each user. The key advantage of Colab is the support of the Tesla K80 GPU accelerator. Gensim[13] 3.6.0 is used to load the pre-trained word vector file. Keras[14] 2.2.5 is run on the TensorFlow backend. It enables fast experimentation of deep learning models and can be run on both CPU and GPU. Large size data such as pre-trained word vector files can be stored in Google drive to reduce file uploading time. The sklearn.metrics[15] 0.22.1 is used to evaluate the classification performance of the models.

D. Experimental Result

The performance of the proposed model, CNN-LSTM model is compared with comparison models described in section 5.2 as listed in Table III. In this paper, the experiment is performed on Myanmar news articles text data that contains five topics and the detail of the dataset is listed in Table II. The experiment contains three main parts pre-processing, word embedding and text classification. In the pre-processing phase, we remove unnecessary characters and segment words by BPE tokenizer. Word embedding matrix is constructed by a pre-trained model via the Gensim library and the CNN, RNN, CNN-RNN and CNN-LSTM models are trained by using Keras, model-level library. All models are trained using 10 epochs, 16 batch size, 0.5 dropout rate, 0.01 l2 bias and kernel regularizer, Adam

[4] https://7daydaily.com/
[5] http://burmese.dvb.no/
[6] http://thevoicemyanmar.com/
[7] http://www.thithtoolwin.com/
[8] https://my.wikipedia.org/wiki/
[9] https://www.rabbit-converter.org/
[10] https://en.wikipedia.org/wiki/Recurrent_neural_network
[11] https://colab.research.google.com/
[12] https://jupyter.org/install.html
[13] https://pypi.org/project/gensim/3.6.0/
[14] https://keras.io/
[15] https://scikit-learn.org/stable/

optimizer and binary_crossentropy loss function. The performance of each model is measured with scikit-learn's classification metrics that report precision, recall, f1-score measured on each class and the highest scores on each evaluation metrics precision, recall and f1-score are highlighted in bold. According to the results of the experiment, a joint CNN-LSTM model performs better in F1-score than the comparison models in all classes. CNN model equally performs better on crime and education domains. The training time for each model is also measured in this paper.

According to the measurement results, the CNN model requires the minimum training time (6 min 1 sec) because only one convolution layer is usedin the experiment. Although the CNN-LSTM model performs better than CNN in three topics in term of F1-score, the CNN model gets at least 4x faster in training time than CNN-LSTM (24 min 5sec) model and at least 3x faster than the remaining models, CNN-RNN (12 min 56 sec) and RNN (13 min 15 sec) model in most datasets. To sum up, the joint CNN-LSTM model outperforms than CNN, RNN and CNN-RNN models in most domains but it requires much training time than other models. The use of simple CNN with one layer convolution faster in training time although the accuracy of the model is slightly degraded in some topics than the CNN-LSTM model.

TABLE III. COMPARISON OF TEXT CLASSIFICATION PERFORMANCE ON FIVE TOPICS

Class	Precision				Recall				F1-score			
	CNN-RNN	RNN	CNN	CNN-LSTM	CNN-RNN	RNN	CNN	CNN-LSTM	CNN-RNN	RNN	CNN	CNN-LSTM
Sport	0.91	0.91	0.88	**0.93**	0.93	0.93	**0.95**	0.94	0.92	0.92	0.92	**0.94**
Art	0.85	0.86	0.87	**0.88**	**0.91**	0.87	0.90	0.90	0.88	0.87	0.88	**0.89**
Crime	0.87	0.86	**0.89**	0.88	0.91	0.92	0.90	**0.93**	0.89	0.89	***0.90***	***0.90***
Health	**0.94**	0.92	0.91	0.91	0.87	0.88	0.89	**0.90**	0.90	0.90	0.90	**0.91**
Education	0.91	0.90	**0.93**	0.91	0.85	0.84	0.84	**0.86**	0.88	0.87	***0.88***	***0.88***

VI. CONCLUSION

This paper performs the comparative experiments on the joint CNN-LSTM model with CNN, RNN and CNN-RNN models on five categories including sport, art, crime, health, and education. We initially experimented with many convolution layers in the CNN model to get higher performance and to catch long term information, but the performance was not increased as expected. Moreover, the max-pooling layer of the CNN model led to the loss of the local context information. So, we use only one convolution layer to extract features and the LSTM layer instead of the max-pooling layer in order to catch long term information and to reduce the loss of context information. According to the experiment, the joint CNN-LSTM model performs better than CNN, RNN, and CNN-RNN models in most domains, but it takes much training time than the remaining models.

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Analysis of Word Vector Representation Techniques with Machine-Learning Classifiers for Sentiment Analysis of Public Facebook Page's Comments in Myanmar Text

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Abstract

This paper presents a study of comparison on three different machine learning techniques to sentiment analysis for Myanmar language. The fundamental part of sentiment analysis (SA) is to extract and identify the subjective information that is social sentiment in the source text. The sentiment class is positive, neutral or negative of a comment. The experiments are done on the collected 10,000 Facebook comments in Myanmar language. The objective of this study is to increase the accuracy of the sentiment identification by using the concept of word embeddings. Word2Vec is used to train for producing high-dimensional word vectors that learns the syntactic and semantic of word. The resulting word vectors train Machine Learning algorithms in the form of classifiers for sentiment identification. This experimental results prove that the use of word embeddings from the collected real world datasets improved the accuracy of sentiments classification and Logistic Regression outperformed the other two ML methods in terms of accuracy and F-measures.

Keywords— *Multiclass classification, natural language processing, sentiment analysis, Facebook Page's comments, word embedding, Logistic Regression.*

I. INTRODUCTION

Sentiment Analysis (SA) is a type of contextual mining of text that identifies and extracts the tendency of people's feelings via Natural Language Processing, computational linguistics and text analysis, which are helped to use extracting and analyzing subjective information from the website majority social media and similar sources. It is used to quantify the general public's social attitude of specific brand, product or service when it monitors online conversations. Sentiment Analysis (SA) is also known as opinion mining. SA uses data mining techniques to extract and capture analyzed data to classify the opinion of a document or collection of documents, like blog posts, reviews of product and social media feeds like Tweets, status updates and comments.

In the distinctive events of technology, people stored data from the internet. These storing data have been grown every day and the large amount of data is stored up to date. But very vital information is carried by these data carry concerning the emotions of different people around the world, so it has become essential to summarize these large number of data with particular automated systems. In this day, many people use social media around the world like Twitter, Facebook and so on. Among of them, Facebook is a popular free social networking website. In Myanmar, most people use Facebook to express their opinions and feelings.

This paper analysis the performance of the word vector techniques (Word2Vec , TFIDF text feature vector representation and pre-trained Word2Vec) with three different machine learning techniques (Logistic Regression, SVM and Random Forest) for public Facebook Page's comments of Myanmar text.

This paper is constructed as the following. The related techniques of word vector is exhibited as Section 2. Section 3 describes the details of proposed system. The experiment results is explained at the Section 4. Finally, Section 5 concludes the discussion with possible enhancements to the proposed system.

II. RELATED TECHNIQUES

A. Word Embeddings

A word embedding learns the text representation where words that have the same meaning have a similar representation. It is emerged

Word Embedding in the Natural Language Processing (NLP) field. Word Embeddings, a text mining technique, is to establish association between words in the set of sentences. The syntactic and semantic meanings of words are comprehend from the context. The idea of distributional assumption propose that words occurring in the alike words are semantically alike. There are the two techniques of word embeddings – (a) Frequency Based Embeddings (b) Prediction Based Embeddings. The Frequency Based Embeddings processed poorly at conserving the contextual information in textual data such as the traditional bag-of-words model. The Prediction Based Embeddings predicts a target word from the given a context word. The researchers developed Global Vectors for Words Representation (GloVe) which is the algorithm of an unsupervised learning to achieve word vector representation does very well at context preservation.

B. Word2Vec

Word2Vec is the producing of the model for word embeddings from a set of sentences for word representation.Word2Vec represents content of vectors into vector space. Word2Vec models use shallow two-layer neural networks, consists of one hidden layer (projection layer) between input and output, which are used to train for reconstructing linguistic contexts of words. Word2Vec takes a set of sentences (corpus) as its input and produces word vectors in a vector space, generally several hundred of dimensions, with each unique word in the text corpus being placed a corresponding vector in the space. Word vectors are placed in the vector space in such a way the words that have similar contexts in the set of sentences are appeared in approximately close to each other in the space. In the same way, different meaning of word contents are located far away from each other. Word2Vec have two techniques to achieve word vectors. They are (i) Continuous Bag of words (C-BOW model) and (ii) Skip-Gram model as shown in Figure 1 and Figure 2. The C-BOW model is used to train for predicting the current word based on its context words. The Skip-Gram model is used to train the model for predicting the context words given a current word. w(t) is the context word and [w(t-2),w(t-1),w(t+1) and w(t+2)] are the surrounding words.

C. tfidf Vectorization

tfidf is an shorthand for term frequency-inverse document frequency. The algorithm of tfidf is a very common algorithm for converting word into a meaningful numeric representation. tfidf vector is based on the concurrency apporach. It is different from the count vectorization approach because it takes into account not just the number of times a text in a single document but in all documents of the corpus. The technique is widely used as features extraction across various applications in NLP field. tfidf need to assign weight for the word based on the number of a word that occurs in the document also consideration the occurrences of the word in all documents.

tf determines the frequency of word appears in a specific document. The calculation equation is as:

$$tf = \frac{\text{occurencies of a word in the document}}{\text{total words in the document}} \quad (1)$$

Inverse Document Frequency for a specific word measures the log of the division of the sum of all document numbers and the document numbers with concluding the particular word in it. It is calculated as the following.

$$idf = log \frac{\text{Total number of docs}}{\text{Number of docs} \in \text{the word}} \quad (2)$$

$$tf - idf = tf * idf \quad (3)$$

tfidf is needed to transform from the specific words to numerical feature vectors.

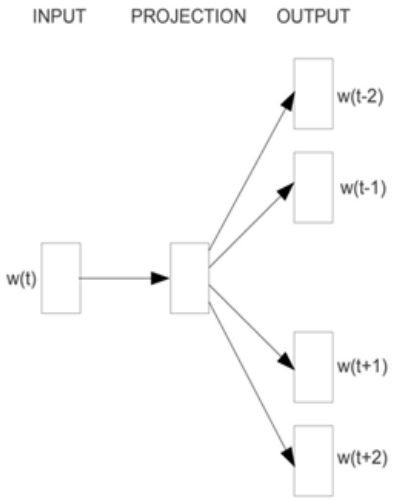

Figure 1. Architecture of C-BOW Model

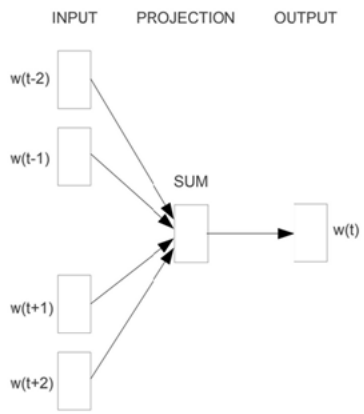

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Figure 2. Architecture of Skip-Gram Model

III. METHODOLOGY

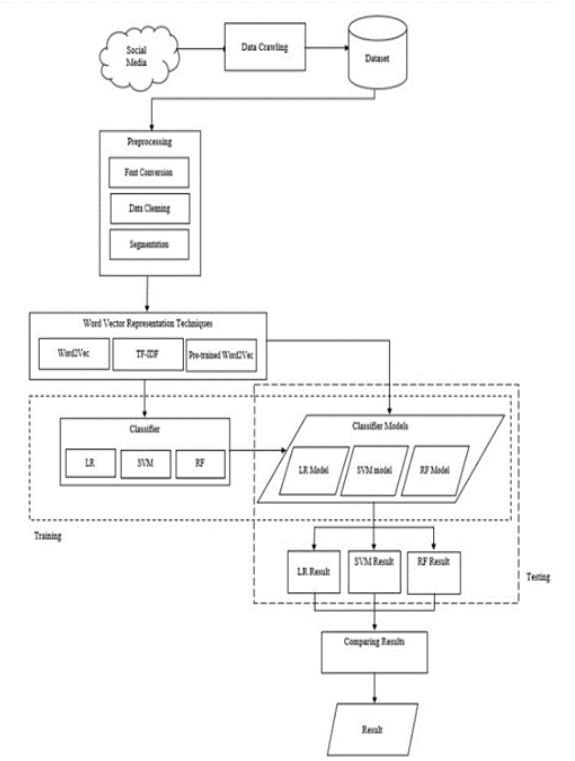

Figure 3. Proposed System Design

The following steps are processed in this proposed system.

A. Data Collecting

There is no previously created dataset for public Facebook page comments in Myanmar language. So, the dataset is crawled public comments of Facebook page related to "Myanmar Celebrity" Page in Myanmar to create own dataset. Social media data (Facebook comments) is collected through data crawling using Facepager tool. These comments were copied and saved as two text file (training text file and testing text file). This corpus is comprised of posts over 100 and comments of 10,000. The set of sentences are antecedently identified with the labels for indicating the user emotions each Facebook comment. The following figure show the visual analysis of imbalance between the three sentiment classes of the dataset. There are 10,000 sentences training dataset in which 2,316 sentences labeled by negative, 1,461 sentences labeled by neutral and 6,223 sentences labeled by positive. There are 1,000 sentences testing data in which 345 sentences labeled by negative, 148 sentences labeled by neutral and 507 sentences labeled by positive.

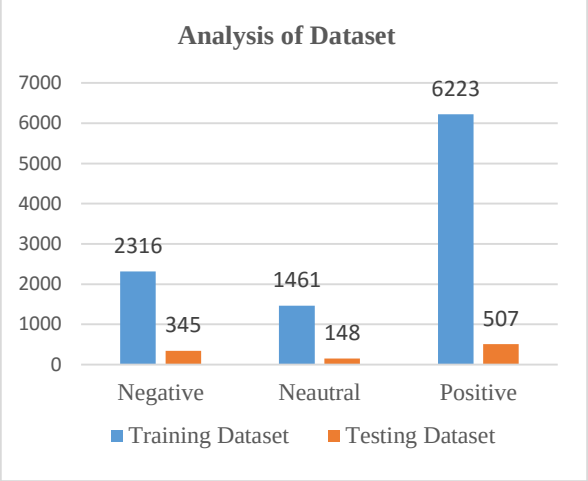

Figure 4. Analysis of Facebook comments Dataset

B. Data Preprocessing

- Font Conversion

  Most of Facebook user in Myanmar use Zawgyi font on social media. Application in technology field use Unicode. In this paper, the collected text data are both Zawgyi and Unicode format. So, the gather text data with Zawgyi format are converted to Unicode by using online Zawgyi-Unicode converter.

- Data Cleaning

  The collected text comments in the dataset consist of various kinds of noises. Before the training and testing are proceeded, these noises are removed all undesirable marks of punctuation, hash sign, additional spaces, unrecognised characters, number etc. from the set of sentences. Most of Facebook users tag the name in the comments so this tagging name is also removed. Due to occur noise in the corpus, the comments which may take place more than one time in the corpus are taken as one.

- Segmentation

  Word segmentation is a necessary part before to process Natural Language Processing in the Myanmar text, because a text of Myanmar language is a string of characters from left to right without explicit word boundary elimiters. It is firstly needed to segment text into separate meaningful words in order to extract words from the collected dataset in Myanmar text. In this paper, Myanmar text data are manually segmented.

TABLE I. EXAMPLE OF SEGMENTATION THE COMMENTS

အမြဲအားပေးနေပါတယ်ချစ်တဲ့မ (I'm always encouraging lovely Ma) အမြဲ အားပေး နေ ပါ တယ် ချစ် တဲ့ မ	Positive
ဘာရေးရမှန်းမသိတော့ဘူး😂😂😂 (I don't know what to write 😂😂😂) ဘာ ရေး ရ မှန်း မ သိ တော့ ဘူး 😂 😂 😂	Neutral
Mcကိုစိတ်ပျက်လာပီ (Mc is disappointed) Mc ကို စိတ်ပျက် လာ ပီ	Negative

C. Word Vector Representation Method

- Building Word2Vec Model

  The study of this paper use C-BOW model of the Word2Vec algorithm. The implementation of Word2Vec in the Gensim python library was used to train on the treated data. For achieving the better implementation of Word Embeddings, most consideration is needed to give certain hyper-parameters. These parameters are: training dataset, dimensionality, context-window, minimum word count, sub-sampling and iteration. The training dataset trained 10,000 sentences of comments (training data). Negative sampling was set because it proved to be efficiently computational relative to hierarchical softmax. For resulting in the better Word Embeddings, the projection layer of the neural network was assigned 300. The context-window size was used 50 to prescribe context-window size for C-BOW model. 1e-3 was used the sub-sampling rate to counter the imbalance dataset of frequent words. 1 set as the minimum count for considering each word in the set of sentences during processing of training. It trained Word2Vec model on the treated data with the above mentioned settings of hyper-parameter and stored in a data file format. The word embedding model produced d*p dimension word vectors where d is the dictionary size and p is the projection layer (hidden layer) size. In this study, the size of the dictionary for the training set is 5,833 vocabulary in the dictionary.

- Building tfidf Vectorization

  tfidf implementation of sklearn python library was trained on the data for training. The same training dataset was used to take as its input corpus. The word embedding model produce word vectors having r×u dimension where r is the number of row in the given training data and u is the unique words from all text. In this study, the rare words is 319 words.

- Building Pre-trained Word2Vec Model

  For pre-trained Word2Vec, the pre-trained C-BOW model of Word2Vec are previously created by using another dataset containing 750,000 sentences. The training data and testing data may or may not contain in this dataset. The pre-trained Word2Vec model is similarly created with the Word2Vec algorithm. In this paper, the 300 dimensions of this model is assigned. The context window of 5 was used for C-BOW models with workers of 10 (the number of processor). And then, the generated pre-trained word vector model is the vocabulary size of 49,740 unique words with 300 dimensions.

D. Classification

In the field of the sentiment analysis, supervised learning is useful for training the data on a pattern that may analyze for either the opinion is positive, negative or neutral. In this paper, we have chosen the three Machine Learning classification techniques consisting of Logistic Regression (LR), Support Vector Machine (SVM) and Random Forest (RF).

IV. EXPERIMENTAL RESULTS

The analysis has been performed comparative according to examine the performance of the three

techniques of classification: Logistic Regression, SVM and Random Forest. In the sentiment classification field, the evaluation of the techniques have used the common information retrieval metrics; precision, recall and f-score. According to evaluate the process of sentence classification, an information retrieval task is considered. The precision can be calculated as (4):

$$Precision = \frac{TP}{(TP+FP)} \quad (4)$$

In which, *TP* is true positive (the occurrences of targets actually identified sentences) and *FP* is false positive (the occurrences of targets that were not correctly identified sentences). Moreover, recall can be calculated as (5):

$$Recall = \frac{TP}{(TP+FN)} \quad (5)$$

In which, *TP* is true positive (the occurrences of targets actually identified sentences) and *FN* is false negative (the occurrences of sentences which were not identified at all). It is possible to calculate the f-score as follows (6):

$$F1 - score = \frac{2*(Precision*Recall)}{(Precision+Recall)} \quad (6)$$

A. Logistic Regression

Logistic Regression classifier is used to apply with the proposed features vectors extraction (tfidf vectorizer, Word2Vec and pre-trained Word2Vec). Using the confusion metrics which are precision (Positive Predictive Value), recall (True Positive Rate) and F-score, the evaluation has been performed. The following tables show the results.

In TABLE II, the evaluation of classification details how the logistic regression classifier with tfidf word vector representation technique carried out for predicting every sentiment label. In the negative label, the classifier exactly prophesied TPR of 38% with the PPV of 63% and F1-score of 47% in supporting 345 negative test objects. In the neutral label, the classifier exactly prophesied TPR of 0.1% with the PPV 44% and F1-score of 16% in supporting 148 neutral test objects. In the positive label, the classifier exactly predict 91% of 507 positive test objects, properly with the PPV of 61% and F1-score of 73%. Logistic Regression classifier with tfidf vectorizer also tried to precisely predict like the other training algorithms. In TABLE III, many objects that were not correctly prophesied label as positive. For instance, according to the Table 3, 203 out of 345 negative comments were predicted to be positive while 92 out of 148 neutral comments were predicted to be positive.

TABLE II. EVALUATION OF LOGISTIC REGRESSION WITH TFIDF VECTORIZER

	PPV	TPR	F1-score	Support
Negative	0.63	0.38	0.47	345
Neutral	0.44	0.1	0.16	148
Positive	0.61	0.92	0.73	507
Average/ Total	0.59	0.61	0.56	1000

TABLE III. CONFUSION MATRIX FOR LOGISTIC REGRESSION WITH TFIDF VECTORIZER

Negative	130	12	203
Neutral	42	14	92
Positive	36	6	465
	Negative	Neutral	positive

In TABLE IV, the evaluation of classification details the Word2Vec+Logistic Regression classifier with carried out for predicting every sentiment class. In the negative label, the classifier exactly prophesied 82% of rate 77% and F1-score of 80%. In the neutral label, the classifier exactly prophesied 38% of 148 neutral test objects, properly with the PPV of 63% and F1-score of 47%. In the positive label, the classifier exactly prophesied 93% of 507 positive test objects, properly with the PPV of 86% and F1-score of 89%. This approach also tried to fit for precisely classifying like other training algorithms. In TABLE V, 44 out of 345 negative comments were predicted to be positive while 57 out of 148 neutral comments were predicted to be negative.

TABLE IV. EVALUATION OF LOGISTIC REGRESSION WITH WORD2VEC

	PPV	TPR	F1-score	Support
Negative	0.788	0.82	0.804	345
Neutral	0.629	0.378	0.473	148
Positive	0.857	0.933	0.893	507
Average/ Total	0.80	0.81	0.80	1000

TABLE V. CONFUSION MATRIX FOR LOGISTIC REGRESSION WITH WORD2VEC

Negative	283	18	44
Neutral	57	56	35
Positive	19	15	473
	Negative	Neutral	Positive

In TABLE VI, the evaluation of classification details how Pre-trained Word2Vec word vector representation technique + the Logistic Regression classifier carried out to predict every sentiment class. In the negative label, the classifier exactly prophesied 77% of 345 negative test objects properly with the PPV of 77% and F1-score of 77%. For the neutral class, the classifier exactly prophesied 38% of 148 neutral test objects properly with the PPV of 53% and F1-score of 44%. For the positive class, the classifier exactly prophesied 92% of 507 positive test objects properly with the PPV of 85% and F1-score of 88%.

TABELE VI. EVALUATION OF LOGISTIC REGRESSION WITH PRE-TRAINED WORD2VEC

	PPV	TPR	F1-score	Support
Negative	0.765	0.774	0.769	345
Neutral	0.533	0.378	0.443	148
Positive	0.85	0.915	0.881	507
Average/ Total	0.77	0.79	0.78	1000

TABLE VII. CONFUSION MATRIX FOR LOGISTIC REGRESSION WITH PRE-TRAINED WORD2VEC

Negative	267	33	45
Neutral	55	56	37
Positive	27	16	464
	Negative	Neutral	Positive

B. Support Vector Machine (SVM)

SVM classifier is applied with the tfidf vectorizer, Word2Vec and pre-trained Word2Vec. Using the confusion metrics which are precision (Positive Predictive Value), recall (True Positive Rate) and F-score, the evaluation has been performed. The following tables show the results.

In TABLE VIII, the evaluation of classification details how the SVM classifier with tfidf word vector representation technique carried out for each sentiment class. In the negative label, the classifier exactly prophesied 37% of 345 negative test objects properly with 60% of the PPV and F1-score of 46%. In the neutral label, the classifier definitely prophesied 0.4% of 148 neutral test objects correctly with 35% of the PPVand F1-score of 0.7%. In the positive label, the classifier exactly prophesied 91% of 507 positive test objects correctly with the PPVof 60% and F1-score of 72%. Logistic Regression classifier with tfidf vectorizer also struggled to precisely classify like other training algorithms. As shown in TABLE IX, many objects that were not correctly prophesied were labeled as positive. For instance, in TABLE IX, 212 out of 345 negative comments were predicted to be positive while 99 out of 148 neutral comments were predicted to be positive.

TABLE VIII. EVALUATION OF SVM WITH TFIDF VECTORIZER

	PPV	TPR	F1-score	Support
Negative	0.604	0.371	0.46	345
Neutral	0.353	0.041	0.073	148
Positive	0.597	0.907	0.72	507
Average/ Total	0.56	0.59	0.53	1000

TABLE IX. CONFUSION MATRIX SVM WITH IFIDF VECTORIZER

Negative	128	5	212
Neutral	43	6	99
Positive	41	6	460
	Negative	Neutral	Positive

In TABLE X, the evaluation of classification details how the SVM classifier with Word2Vec word vector representation technique processed for predicting every sentiment label. In the negative label, the classifier definitely prophesied 82% of 345 negative test objects properly with the PPV of 78% and F1-score of 80%. In the neutral label, the classifier definitely prophesied 34% of 148 neutral test objects properly with the PPV of 56% and F1-score of 42%. In the positive label, the classifier definitely prophesied 92% of 507 positive test objects correctly

with the PPV of 85% and F1-score of 89%. SVM classifier with Word2Vec also struggled to precisely classify like other training algorithms. According to the TABLE XI, 42 out of 345 negative comments were prophesied to be positive while 60 out of 148 neutral comments were prophesied to be negative.

TABLE X. EVALUATION OF SVM WITH WORD2VEC

	PPV	TPR	F1-score	Support
Negative	0.775	0.817	0.795	345
Neutral	0.556	0.338	0.42	148
Positive	0.853	0.919	0.885	507
Average/ Total	0.78	0.80	0.79	1000

TABLE XI. CONFUSION MATRIX FOR SVM WITH WORD2VEC

Negative	282	21	42
Neutral	60	50	38
Positive	22	19	466
	Negative	Neutral	Positive

In TABLE XII, the evaluation of classification details how the SVM classifier with Pre-trained Word2Vec word vector representation technique carried out predicting for every sentiment label. In the negative label, the classifier definitely prophesied 80% of 345 negative test objects correctly with the PPV of 75% and F1-score of 77%. In the neutral label, the classifier definitely prophesied 32% of 148 neutral test objects properly with the PPV of 53% and F1-score of 40%. In the positive label, the classifier definitely prophesied 92% of 507 positive test objects properly with the 85% of PPV and F1-score of 88%.

TABLE XII. EVALUATION OF SVM WITH PRE-TRAINED WORD2VEC

	PPV	TPR	F1-score	Support
Negative	0.751	0.797	0.774	345
Neutral	0.533	0.324	0.403	148
Positive	0.853	0.915	0.883	507
Average/ Total	0.77	0.79	0.77	1000

TABLE XIII. CONFUSION MATRIX FOR SVM WITH PRE-TRAINED WORD2VEC

Negative	275	26	44
Neutral	64	48	36
Positive	27	16	464
	Negative	Neutral	Positive

C. Random Forest

Random Forest classifier is applied with the proposed features extraction (tfidf vectorizer, Word2Vec and pre-trained Word2Vec). Using the confusion metrics which are precision (Positive Predictive Value), recall (True Positive Rate) and F-score, the evaluation has been performed. The following tables show the results.

In TABLE XIV, the evaluation of classification details how the Random Forest classifier with tfidf word vector representation technique carried out for every sentiment label. In the negative label, the classifier definitely prophesied 36% of 345 negative test objects properly with the PPV of 60% and F1-score of 45%. In the neutral label, the classifier definitely prophesied 10% of 148 neutral test objects properly with the PPV of 26% and F1-score of 15 %. In the positive label, the classifier definitely prophesied 90% of 507 positive test objects properly with the 62% of PPV and F1-score of 73%. Logistic Regression classifier with tfidf vectorizer also struggled to precisely classify like other training algorithms. As TABLE XV, many objects that were not correctly prophesied were labeled as positive. For instance, in TABLE XV, 203 out of 345 negative comments were prophesied to be positive while 92 out of 148 neutral comments were prophesied to be positive.

TABLE XIV. EVALUATION OF RANDOM FOREST WITH TFIDF VECTORIZER

	PPV	TPR	F1-score	Support
Negative	0.6	0.357	0.447	345
Neutral	0.263	0.101	0.146	148
Positive	0.617	0.9	0.731	507
Average/ Total	0.56	0.59	0.55	1000

TABLE XV. CONFUSION MATRIX FOR RANDOM FOREST WITH TFIDF VECTORIZER

Negative	130	12	203
Neutral	42	14	92
Positive	36	6	465
	Negative	Neutral	Positive

In TABLE XVI, the evaluation of classification details how the Random Forest classifier with Word2Vec word vector representation technique carried out for every sentiment label. In the negative label, the classifier definitely prophesied 78% of 345 negative test objects correctly with the PPV of 72% and F1-score of 75%. In the neutral label, the classifier definitely prophesied 12% of 148 neutral test objects properly with the PPV of 43% and F1-score of 20%. In the positive label, the classifier definitely prophesied 93% of 507 positive test objects properly with the PPV of 81% and F1-score of 93%. Random Forest classifier with Word2Vec also struggled to precisely classify like other training algorithms. According to TABLE XVII, 58 out of 345 negative comments were prophesied to be positive while 76 out of 148 neutral comments were prophesied to be negative.

TABLE XVI. EVALUATION OF RANDOM FOREST WITH WORD2VEC

	PPV	TPR	F1-score	Support
Negative	0.717	0.78	0.747	345
Neutral	0.429	0.122	0.189	148
Positive	0.808	0.929	0.864	507
Average/ Total	0.72	0.76	0.72	1000

TABLE XVII. CONFUSION MATRIX FOR RANDOM FOREST WITH WORD2VEC

Negative	269	18	58
Neutral	76	18	54
Positive	30	6	471
	Negative	Neutral	Positive

In TABLE XVIII, the evaluation of classification details how the Random Forest classifier with Pre-trained Word2Vec word vector representation technique carried out for every sentiment label. In the negative label, the classifier definitely predicted 75% of 345 negative test objects properly with the PPV of 71% and F1-score of 73%. In the neutral label, the classifier definitely prophesied 17% of 148 neutral test objects correctly with the PPV of 46% and F1-score of 24%. In the positive label, the classifier definitely prophesied 92% of 507 positive test objects correctly with the PPV of 80% and F1-score of 86%.

TABLE XVIII. EVALUATION OF RANDOM FOREST WITH PRE-TRAINED WORD2VEC

	PPV	TPR	F1-score	Support
Negative	0.707	0.748	0.727	345
Neutral	0.463	0.169	0.2448	148
Positive	0.8	0.917	0.855	507
Average/ Total	0.72	0.75	0.72	1000

TABLE XIX. CONFUSION MATRIX FOR RANDOM FOREST WITH PRE-TRAINED WORD2VEC

Negative	258	20	67
Neutral	74	25	49
Positive	33	9	465
	Negative	Neutral	Positive

V. CONCLUSION

The paper of this study compares the performance of 3 different Machine Learning (ML) techniques containing Logistic Regression, SVM and Random Forest while using Myanmar sentiment analysis based on word vector representation method. The dataset that has been used is a collection of Myanmar Facebook pages comments for the purposes of sentiment analysis. The experimental results have proved that Logistic Regression classifier with Word2Vec has exceeded in performance than the other two Machine Learning by obtaining **80%** of F1-score. Hence, we can draw to conclude that given a word vector representation from Word2Vec instead of tfidf and pre-trained Word2Vec for extracting feature vectors, sentiment analysis in Myanmar on three labels of ideas, will carried out better the use of Logistic Regression than the use of SVM and Random Forest.

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Building Speaker Identification Dataset for Noisy Conditions

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Abstract

Speech signal processing plays a crucial role in any speech-related system whether Automatic Speech Recognition or Speaker Recognition or Speech Synthesis or something else. Burmese language can be considered as an under resourced language due to its linguistic resource availability. For building Burmese speaker identification system, the sufficient amount of speech data collection is a very challenging task in a short time. In order to get higher data size, this paper analyzes that the getting higher duration of speech data actually combining with various noises encountering in our surroundings. For increased noisy state speech data, we also used the voice activity detection (VAD) technique to acquire only the speaker specific information. For feature extraction, we used MFCC, Filter Banks and PLP techniques. The experiments were developed with i-vector methods on GMM-UBM together with PLDA and presented the performance of different data set in the form of EER with two models trained on clean and noisy data to prove that the developed speaker identification system is noise robust.

Keywords— *Burmese Speaker Identification, noise robustness, VAD, MFCC, Filter Banks, PLP, GMM-UBM, PLDA*

I. INTRODUCTION

The speech of living things especially for human involves numerous discriminative acoustic features that can be discerned who they are because of the structural formation of vocal tract is unique for everyone. Speaker identification is the process by which the acoustic speech signals to its corresponding speaker and is applied in many applicable areas. Speech corpora collection is the very first step in building speaker identification system. In order to develop the speaker identification system, the sufficient speech data are needed to train and test the spoken speech data. The performance of the system is also depended on the amount of speech data. There are many variations in speaker identification system. The first one is the duration of utterances. The longer the utterances, the better recognizes the corresponding speaker. A second variation is noise as any kinds of noise make the identification process harder. The third variation is accent or speaker specific facts. The speaker is easier to identify if he/she speaks a standard dialect or the ones that matches the speech data the system trained on. The final variation is the speech recorded conditions. Therefore, we will propose Burmese speaker identification how to construct with noisy data. The paper is organized as follows. Related works will be presented in section II. In section III, speaker identification process will be introduced. Section IV will be described the types of speaker recognition and the proposed architecture of speaker identification system with noisy data will be expressed in section V. Experimental setup will be addressed in section VI and experimental results will be discussed in section VII. Finally, conclusion will be expressed in section VIII.

II. RELATED WORKS

There are many speaker recognitions with various approaches found in publications.

Arnab Poddar, Md Sahidullah and Goutam Saha [2] presented the comparison of two different speaker recognition systems, i-vector based and GMM_UBM in utterance duration variability. It revealed that GMM_UBM system outperforms i-vector system for very short test utterances if the speaker are enrolled with sufficient amount of training data whereas total variability (i-vector) based system degrades with the reduction in test utterance length and also require huge computational resource development data for identifying the speaker although GMM_UBM don't require the huge amount of development data.

Comparison of text independent speaker identification systems using GMM and i-vector

methods are done by Nayana P.K., Dominic Mathew, and Abraham Thomas [3]. It was observed that appending formants and pitch high level features to basic features: PNCC (Power Normalized Cepstral Coefficients) and RASTA PLP (Relative Spectral PLP) obtain the better accuracy for speaker identification. It was also showed that the accuracy of i-vector method with PLDA classifier is better than that of Cosine Distance Scoring (CDS) classifier. Moreover, it revealed that the system performance enhances when longer utterances are used.

Analysis of various feature extraction techniques for robust speaker recognition was presented by Qin Jin and Thomas Fang Zheng [4] to help the researchers for catching the current front end features classified as low level and high level features. They surveyed the speaker recognition system on different feature extraction techniques: MFCC, MVDR, FDLP, MHEC, SCF/SCM, FFV, HSCC and Multitaper MFCC and presented the strength and weakness of these techniques.

R.ARUL JOTHI M.E [5] presented the analysis of suitable extraction methods and classifiers for speaker identification since 2017. It identified the speaker's voice whether original or disguised voice based on MFCC, Delta MFCC and Delta-Delta MFCC with SVM classifier. MFCC with SVM classifier improves the performance of system and accuracy rates up.

An improved approach for text independent speaker recognition was proposed by Rania Chakroun [6]. It proposed that the new feature extraction method combining MFCC and Short Time Zero Crossing Rate (ZCR) of the signal. ZCR is the number of times the zero axes crossed by the signal per frame. By comparing the performance of two speaker recognition systems with the use of MFCC and combination of MFCC and STZCR, it showed the new proposed feature extraction yields better outcome and reduced in EER.

III. SPEAKER IDENTIFICATION

Speaker identification determines the speaker identity from which of the registered speakers a given utterance comes. It is a very challenging task because human speech signals are highly variable due to various speaker characteristics, different speaking styles, environmental noises, and so on. There exist various feature extraction methods and approaches for speaker identification system. There are three main steps in speaker identification.

As part of feature extraction, a set of feature vectors are obtained from the raw speech signal to more emphasize the speaker related information because the speech signal can contain many features which are not required to claim the speaker. Therefore, feature extraction process is advantageous when you need to diminish the resources required for processing without losing relevant information. There are many different types of features that can be extracted. The recognition accuracy rate varies according to chosen extraction methods. Various types of available features are high level features, spectra-temporal features, short term (low level) spectral features, and prosodic features [3]. In these, low level features are easy to extract and very effective to recognize the speaker. Although high level features contain more speaker specific information, the extraction process is more complicated.

To generate the speaker models representing to each speaker, the features attained from the feature extraction stage are used and stored these speaker models into a database as UBM for performing the comparison during testing. It is the main session of speaker identification system as the models created in this stage are applied to perform comparison in the identification stage. Different modeling methods are HMM (Hidden Markov Models), GMM (Gaussian Mixture Models), DNN (Deep Neural Network), and i-vector method.

Identifying the test speech signal is the final stage of every speaker identification system. Relative scores corresponding to each of the speaker models are computed and then the one which has the highest score is identified as the target speaker. Different scoring methods used for identification are CDS (Cosine Distance Scoring), PLDA (Probabilistic Linear Discriminant Analysis), LLR (Log Likelihood Ratio), and SVM (Support Vector Machine) and so on.

IV. TYPES OF SPEAKER RECOGNITION

There are two types of speaker recognition: speaker verification and speaker identification. If the speaker claims to be of a certain identity and the voice is used to verify this claim, this is called verification or authentication. Speaker verification is a 1:1 match where one speaker's voice is matched to one template (also called a "voice print" or "voice model"). Speaker identification is the task of determining an unknown speaker's identity. Therefore, it is a 1: N match where the voice is compared against N templates. It involves

two phases: enrollment and testing. During enrollment, the speaker's voice is recorded and typically a number of features are extracted to form a voice print, template, or model. In testing phase, a speech sample or "utterance" is compared against a previously created voice print. Moreover, there exist two types of speaker identification: text dependent and text independent. Text dependent speaker identification needs to utter exactly the same utterance to determine who they are. Text independent speaker identification has no limits and constraints on the spoken words that are uttered. It is more flexible and usable in real world applications. Verification is faster than identification because of the processing time of matching. This paper proposes text independent speaker dependent identification because text independent systems are most applicable in real world.

V. Proposed Architecture Of Speaker Identification System

Feature extraction, speaker modeling and identification process are the three crucial stages of any speaker recognition system. This section presents the detail description of the whole speaker identification process. The proposed architecture of speaker identification system with noisy data exhibits in Fig. 1.

A. Data Preprocessing

Before feature extraction, we need to firstly preprocess the data. In this stage, the whole recorded audio speech data is chopped into utterance level speech segments with ***Audacity*** which is open source and cross-platform audio multi-track audio editor and recorder software. And then, the preprocessed speech data are the utterance level speech segmented data with 16 bits mono PCM in 16 kHz ranging from 10 to 27 seconds. After that, we randomly added the utterance level segmented speech data with various noises found in our surrounding prepared by ourselves. By contaminating, our data set size increases the duration than that of original clean data set size. And then, the original clean data and noise-combined data are combined to use in feature extraction. Surrounding noises include car's horn, fly buzz, the dog's bark, fire alarm, ringtone, cat meow, whistle, roar, shouting, wind blowing, birds chirping, banging of hammer, school bell, beating a drum and so on.

B. Feature Extraction and Voice Activity Detection

There is no standard rule for choosing among these features for the question 'Which feature extraction technique should one use?' It depends on our needs like intended application, robustness, computing resources and amount of data available. Because short-term spectral (low level) features are easy to compute and provide good results, exploring with these types of feature enhances the system performance. Feature extraction stage is one of the most important components in any SR systems and its objective is to find robust and discriminative features in acoustic data because better features give the more improved recognition rate. In our proposed system, clean and noisy data are combined to extract the features. And then, Voice activity detection (VAD) is applied for noisy data. It is a technique used to detect the speech or non-speech section in recorded speech data with the aim of removing the silence frames in segmented speech data, saving the computing time and enhancing the recognition accuracy rate. It also refers to the problem of distinguishing speech segments from background noise in an audio stream and is also language independent. Moreover, we exploited with three kinds of low level feature extraction techniques for system performance: Mel Frequency Cepstral Coefficient (MFCC), Filter Bank and Perceptual Linear Prediction (PLP). The system's recognition rate diverges depending on our choice because there are no standard rules for choosing among these features. It depends on our destined needs in related applied areas.

C. Building Speaker Models

The feature vectors extracted in the feature extraction stage take to build the speaker models. UBM is the key element of an i-vector (existing in low dimensional spaces that are smaller in size to reduce the recognizing time) system as it is necessary for collecting statistics from speech utterances. It is constructed using feature values of sound speech samples from the different speakers and Maximum A Posteriori (MAP) is used to get the speaker models each [1]. It is the central part of this system because it is used in comparison with the test speech segment's feature vector for describing who the speaker is. In this paper, we implement the speaker identification system with i-vector method by using Kaldi ASR open source toolkit to build the speaker models:

Model in Clean Data (Model_1) and Model in Noisy Data (Model_2) [8].

D. Identification Process

Different scoring methods: support vector machine (SVM), Probabilistic Linear Discriminative Analysis (PLDA), and Cosine Distance Scoring (CDS) for identifying the speaker are applied in any speaker identification system. In this paper, i-vector based speaker identification with PLDA is put to work for recognizing the noisy test speech input signal in the sense of three feature extraction techniques on two speaker models: Model_1 and Model_2 for verifying our proposed system, Model_2 is noise robust. PLDA method used in this paper is the simplified or Gaussian PLDA with 200 Gaussian components of 100 dimensions in i-vectors. It is computed the similarity scores as the ratio of the probability that both test and reference i-vector belong to the same speaker to the probability that they both belong to different speakers.

VI. EXPERIMENTAL SETUP

A. Data Preparation

The experiment is implemented with total number of 37 speakers including 12 male and 25 female speakers. There are about 7 hours of clean data comprising of 2516 utterances in clean data and 16 hours of noisy training data set comprising of 5032 utterances in noisy data. It has double the size of data than the clean data. The development data are about 54 minutes in the clean data with 321 utterances and nearly 2 hours of noisy data with 642 utterances. In this paper, we will do the experiments with two test sets: TestSet1 and TestSet2. For TestSet1 and TestSet2, there exist 111 utterances of clean and noisy test sets with the length of 18 seconds and 23 seconds each. These are evaluated based on Model_1 and Model_2 for approving the noise robustness of the model trained with noisy data. Speech data utterances were recorded at 16 bits mono PCM in 16 kHz with the duration of ranging from 10 to 27 seconds each. This frequency rate affects in the feature extraction process and building the speaker models because this rate is suitable for Myanmar's spoken speech tone. The total number of speakers included in this experiment is shown in table I. Data preparation for Model_1 and Model_2 is shown in table II. The experiment using clean data for Model_1 is taken from [7]. Moreover, for Model_2, we randomly reckon additional noise to the original clean data. Test case preparation for TestSet1 and TestSet2 is shown in table III. TestSet1 is one which contains original clean test data and TestSet2 is the test data that randomly combines additional noise to the original clean data.

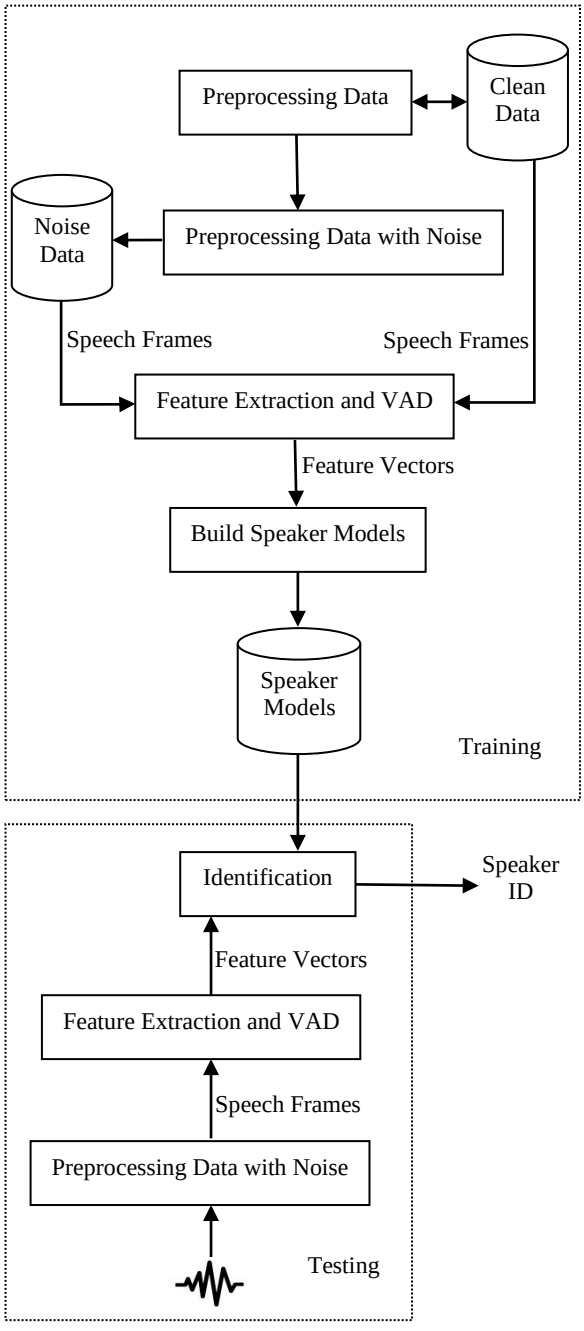

Figure 1. Proposed architecture of noise robust speaker identification system

B. Evaluation: Equal Error Rate(EER)

We appraised automatic evaluation of equal error rate (EER) for assessing the performance of speaker identification models in both conditions. False

acceptance rate (FAR) (1), is a type of error allowing the impostor speaker is improperly identified as the known speaker and false rejection rate (FRR) (2) is incorrectly denied the actual speaker known by the system as impostor. Equal Error Rate (EER) is one where FAR equals to FRR and also the point where FAR and FRR are optimal and minimal. EER of speaker model is mainly based on the amount of training data. Therefore, we are going to collect the speech data more and more in the future. The lower the EER value, the higher the recognizing rate of speaker models.

TABLE I. TOTAL NUMBER OF SPEAKERS

Male	Female	Total Number of Speakers
12	25	37

TABLE II. DATA PREPARATION FOR THE EXPERIMENTS

Data	Number of Utterances		Duration (hr: min: sec)	
	Model_1	*Model _2*	*Model _1*	*Model _2*
Train	2516	5032	07:10:39	16:05:04
Dev	321	642	00:54:36	01:58:35

TABLE III. STATISTICS OF THE TEST SETS

Test Sets	Number of Utterances	Duration (hr: min: sec)
TestSet1	111	00:18:26
TestSet2	111	00:23:12

$$FAR = \frac{Total\ False\ Acceptance}{Total\ False\ Attempts} \quad (1)$$

$$FAR = \frac{Total\ False\ Rejection}{Total\ True\ Attempts} \quad (2)$$

C. *Evaluation:Test Samples' Accuracy*

To evaluate the performance of every test speech samples, we also applied the automatic evaluation shown in (3). This automatic evaluation is based on how many test speech samples recognized by the speaker models differs from the correct test speech samples.

*Accuracy (%) = ((TTSs – WDSs) / TTSs) * 100* (3)

where, *Accuracy*=Test Case's Accuracy in Percentage

TTSs=Total Test Speech Samples in Test Case

WDSs = Wrong Detected Samples .

VII. EXPERIMENTAL RESULTS

We will describe the experimental results based on the models built in Model_1 and Model_2 using PLDA identification. Table IV shows the performance of speaker models, Model_1 and Model_2 on the development data, and two test sets. To evaluate the performance of models, TestSet1 and TestSet2 is used for assessments in terms of equal error rate (EER%). TestSet1 is the original clean test data and TestSet2 is the data prepared with additional noise to the clean data. The performance of speaker models in varying surrounding conditions on the development data sets depicts with a chart in Fig. 2.

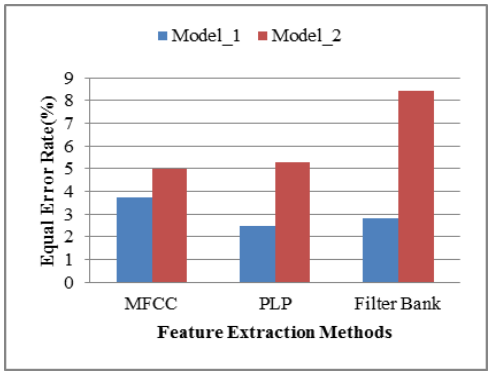

Figure 2. Performance of EER on building speaker models

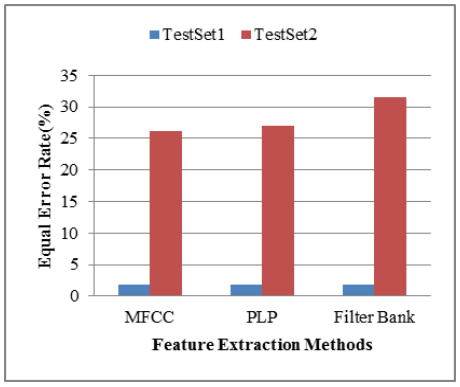

Figure 3. EER of TestSet1 and TestSet2 on Model_1

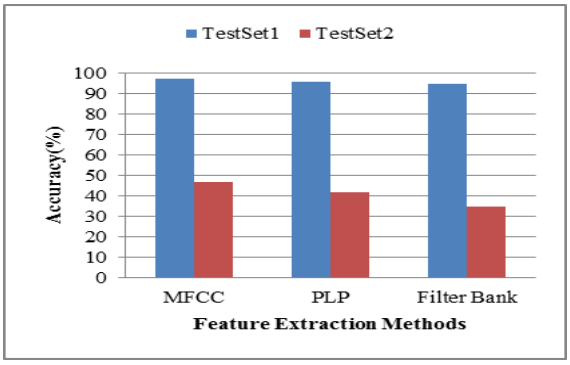

Figure 4. Performance of Model_1 with three feature extraction methods

Figure 3 and Figure 4 reveals that the evaluation results of TestSet1 and TestSet2 on speaker models built in clean data. Although the EERs and accuracies of models show good results on clean test data, the performance degrades on noisy test data. This experiment approves the noisy data needs to be trained on the noisy data.

Table IV shows the performance of two models in terms of EERs on two test sets: TestSet1 and TestSet2. It can be seen that the EER of clean data, TestSet1 has got comparable results in both clean and noisy models. It means, the noisy model can give the similar performance on clean and noisy test data. EER of noisy model, Model_2, was decreased significantly on TestSet2, noisy data, by all feature extraction techniques at most 20.72% than clean model, showing the noisy training data are important for noisy test data. From the experiments, we found MFCC give better results among three feature extraction techniques for clean and noisy conditions of our data.

TABLE IV. PERFORMANCE OF TESTSET1 AND TESTSET2 ON MODEL_1 AND MODEL_2

Feature Extraction Methods	Equal Error Rate (%)					
	Dev		*TestSet1*		*TestSet2*	
	Model_1	*Model_2*	*Model_1*	*Model_2*	*Model_1*	*Model_2*
MFCC	3.738	4.984	1.802	1.802	26.13	8.108
PLP	2.492	5.296	1.802	2.703	27.03	10.81
Filter Bank	2.812	8.424	1.802	2.703	31.53	10.81

TABLE V. ACCURACIES ON TESTSETS (%)

Feature Extraction Methods	Accuracy Rate (%)			
	Model_1		*Model_2*	
	TestSet1	*TestSet2*	*TestSet1*	*TestSet2*
MFCC	97.27	46.36	93.64	76.36
PLP	95.45	41.82	92.73	75.45
Filter Bank	94.55	34.55	94.55	76.36

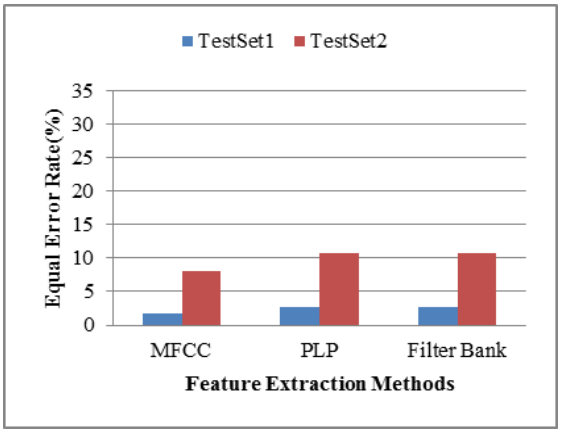

Figure 5. EER of TestSet1 and TestSet2 on Model_2

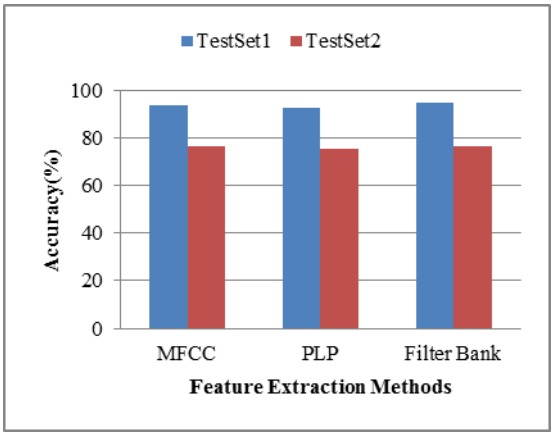

Figure 6. Performance of Model_2 with three feature extraction methods

Fig. 5 and Fig. 6 show that the results of TestSet1 and TestSet2 on noisy speaker models. This paper approaches to the point of view of every noise aggravates the recognition rate on every speech-related processing. The error rates of clean data are sharply higher than that of testing on noisy data.

As Fig. 3, Fig. 4, Fig. 5, Fig. 6, Fig. 7 and Fig. 8 show, it can be seen clearly that the model with noisy data is better than Model_1 in noisy condition and noisy data helps to improve the performance of

speaker identification in both clean and noisy condition. According to the Table IV and V, TestSet2 of on Model_1 and Model_2, show the improved equal error rate on our model, Model_2. The system performance degrades in clean data when testing with noisy test speech data but Model_2 yields satisfiable results on both conditions clean and noisy. In this analysis, we showed the error rates are obviously decreased almost one-third by Model_2 compared to Model_1.

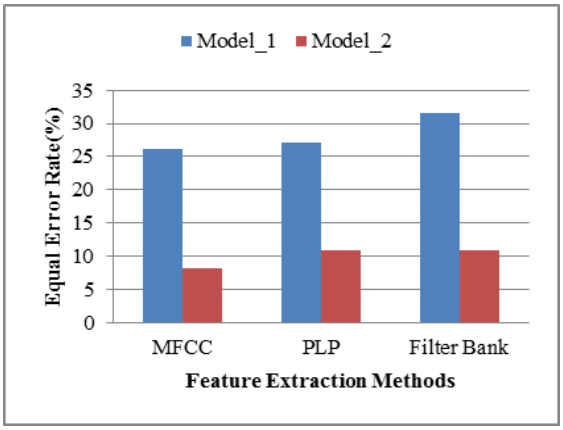

Figure 7. EER on two models with TestSet2

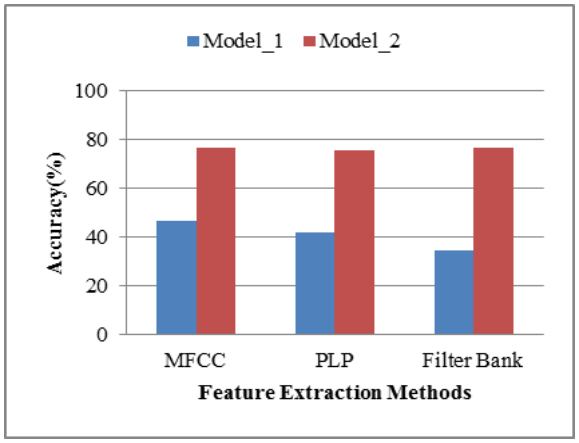

Figure 8. Performance of TestSet2 on two models with three feature extraction methods

VIII. CONCLUSION

Noise delivers detriments in speech-related processing. This paper shows the importance of noisy data preparation for speaker identification. Experiments were done on speaker identification training on clean and noisy data. We also analyzed the rate of change of EER with three feature extraction methods in the experiments. From the experiments, applying MFCC features gave the best results among three different feature extraction techniques. We also found that, integrating additional noise to the original clean data improves the recognizing rate in every feature extraction method with increasing the size of data. From the experiments, it is clear that the preparation of noisy data is effective for noise robust speaker identification system and the results are acceptable.

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English-Myanmar (Burmese) Phrase-Based SMT with One-to-One and One-to-Multiple Translations Corpora

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Abstract

This paper contributes the first investigation of machine translation (MT) performance differences between Myanmar and English languages with the use of several possible Myanmar translations for the specific primary educational domain. We also developed both one-to-one and many Myanmar translations corpora (over 8K and 46K sentences) based on old and new English textbooks (including Grade 1 to 3) which are published by the Ministry of Education. Our developing parallel corpora were used for phrase-based statistical machine translation (PBSMT) which is the de facto standard of statistical machine translation. We measured machine translation performance differences among one-to-many English to Myanmar translation corpora. The differences range between 19.68 and 52.38 BLEU scores from English to Myanmar and between 50.17 and 75.12 BLEU scores from Myanmar to English translation. We expect this study can be applied in Myanmar-to-English automatic speech recognition (ASR) development for primary English textbooks. The main purpose is to translate primary English textbooks data correctly even if the children use in several Myanmar conversation styles.

Keywords—*Phrase-based Statistical Machine Translation (PBSMT), One-to-Many Parallel Corpus, Myanmar-English Machine Translation, Primary English Textbooks of Myanmar, Word Error Rate (WER).*

I. INTRODUCTION

Nowadays, our Myanmar children in rural and less developing areas are facing many difficulties in education. One of the challenges in their educational lives is that English is taught as a second language from kindergarten. The main point for continuous education is to catch up with the knowledge of English since childhood. However, there are no enough of teaching staffs and teaching aid devices. Consequently, most of the primary students in these rural and less developing areas are weak in learning English. To address these challenges, the machine translation technologies can be applied to be more interest in their lessons and to support as learning assistance tool. Although English to Myanmar as in bi-directional translation systems have been successfully used in other domains such as travel, to the best of our knowledge, there are no prior researches targeting the application of machine translation to the education sector. This paper contributes to the first studying of the machine translation performance between Myanmar primary educational textbooks sentences and translated Myanmar sentences by applying statistical machine translation (SMT) approaches (especially PBSMT) with one-to-one and one-to-many translated parallel data. One more contribution is that we are developing a parallel corpus of Myanmar-English on the primary educational domain. We also consider one-to-many possible manual translations for English to Myanmar translation direction.

The structure of the paper is as follows. In the next section, we present a brief review of machine translation systems for Myanmar-English as bi-directional systems. Section III presents the current state of English education at Primary schools in Myanmar and section IV describes the Myanmar-English parallel corpus building for machine translation experiments. In section V, we describe PBSMT as the experimental methodology used in machine translation experiments. Section VI presents the statistical information of the corpus and the experimental settings. Section VII reports the experimental results with discussions and section VIII presents the error analysis on translated outputs. Finally, section IX presents conclusion and future work.

II. RELATED WORK

This section reviews the previous works in statistical machine translation between Myanmar and English languages. To date, there have been some studies on the SMT of Myanmar language.

Thet Thet Zin et al. (2011) [1] described statistical Myanmar phrase translation system with morphological analysis. Here, the experiments were conducted based on total data size of 13,042: 12,827 parallel sentences of which were training set and the rest of 215 were test set. And Bayes' rule was used to reformulate the translation probability for translating Myanmar phrases into English phrases. The evaluation criteria of machine translation were precision, recall and the F-measure. There were problems with many out of vocabulary (OOV) words such as proper noun, noun and verb phrases in the first baseline system. As the second step, the morphological analysis is applied on pre-processing phrase of translation process to address the above OOV problem. According to the results, the morphological analysis method achieved the good comparison with the baseline. However, there were still most errors of post-positional markers that made ambiguous meaning. Therefore, one way of helping that problem, part-of-speech (POS) tagging technique, was applied. Adding morphology and POS of Myanmar language to baseline system gave the best results and reduced OOV rates. But, there were still 95 errors in 215 tested sentences such as unknown foreign words, translation failure, segmentation error, detecting verb phrases error, untranslatable phrases and missing English particles.

Ye Kyaw Thu et al. (2016) [2] presented the first large-scale study of the translation of the Myanmar language. There were a total of 40 language pairs in the study that included languages both similar and fundamentally different from Myanmar. In this experiment, 457,249 sentences were used for training, 5,000 sentences for development and 3,000 sentences for evaluation. The results showed that the hierarchical phrase-based SMT (HPBSMT) [3] approach gave the highest translation quality in terms of both the BLEU [4] and RIBES scores [5].

Win Pa Pa et al. (2016) [6] presented the first comparative study of five major machine translation approaches applied to low-resource languages. PBSMT, HPBSMT, tree-to-string (T2S), string-to-tree (S2T) and Operation Sequence Model (OSM) translation methods were applied to the translation of limited quantities of travel domain data between English and {Thai, Laos, Myanmar} in both directions. Here, 20,000 sentences were used for training, 500 sentences for development and 300 sentences for evaluation. The experimental results indicated that in terms of adequacy (as measured by BLEU score), the PBSMT approach produced the highest quality translations. Here, the annotated tree is used only for the English language for S2T and T2S experiments. This is because there is no publicly available tree parser for Lao, Myanmar and Thai languages. From their RIBES scores, we noticed that OSM approach achieved the best machine translation performance for Myanmar to English translation.

Rui et al. had developed Neural Machine Translation (NMT) and PBSMT systems with pre-ordering for English-Myanmar in both translation directions. All the provided parallel data for all the targeted translation directions, including the training corpus "ALT" and "UCSY" and the "ALT" dev/test data: 226,500 sentences for training, 1,000 sentences for testing and 900 sentences for evaluation were used. The source English part was pre-ordered before being input into NMT and SMT systems. The results also confirmed the slight positive impact of using pre-ordering in English-Myanmar PBSMT [7].

Based on the experimental results of the previous works, in this paper, the PBSMT experiments were carried out to study the performance variations using PBSMT, one-to-one and many translations corpora.

III. ENGLISH EDUCATION AT PRIMARY SCHOOLS IN MYANMAR

In this section, we would like to present about English education at primary schools in Myanmar. In Myanmar's educational sector, English subject is taught as a second language since primary level. It is very important to catch up with the knowledge of the next higher levels because almost next higher level textbooks are published in English. Nowadays, there are two types of primary level English textbooks in Myanmar such as old primary curriculum and new ones. In old ones, the lessons are in general teaching style and the students are weak in interesting and understanding as a consequence.

Today, a drastic education reform has been implementing by the Ministry of Education in Myanmar. And primary education reform is one of the important topics. In 2014, CREATE Project (The Project for Curriculum Reform for Primary Level of Basic Education in Myanmar) was launched for emerging new primary education textbooks, Teacher's Guide, updating assessment. And this project introduced new primary education to in-service and pre-service teachers. CREATE Project is jointly organized by the Ministry of Education in Myanmar and Japan International Cooperation Agency.

From June 2017 to current 2020, new primary Grade 1, 2 and 3 English textbooks were introduced nationwide. New primary education textbooks are richer and made up of attractive contents that promote students' active learning, considering diversification like gender, ethnicities, disabilities, many pictures and photos that stimulate students' interests of learning, colorful however applying universal color style to be friendly for color-blinded students. There are 36 weeks per year and one period is taken in 40 minutes for all Grades [8]. Grade 1 English textbooks cover for alphabetical letters, numbers, short nouns and Grade 2 English textbooks cover for adjective, verb, greeting sentence and short sentence forms. In Grade 3, long sentences, alternate practice sentences, and usage of question words are covered [9].

IV. PARALLEL CORPUS BUILDING

For Myanmar NLP researchers, there are many difficulties which are arisen from the lack of resources; in particular parallel corpora are scare [10]. Currently, there is no specific parallel corpus that can be used for Myanmar Primary (MP) students who would like to study Myanmar-English and vice versa translation. Therefore, as a first step, we are building a textual parallel MP corpus with the purpose of developing a Machine Translation (MT)-based approach for using technology to assist for MP students in their educational life.

For this purpose, we collected the necessary data as main sentences from old and new English textbooks (including Grade 1 to 3) which are published by the Ministry of Education. Then, we translated them in Myanmar by using "The Khit Thit English-Myanmar Pocket Dictionary", compiled by Khit Thit editorial staff and "Pocket Best Speaking" by Professor Minn Nandar (Dr. Min Tin Mon). As in nature of Myanmar language, we found the fact that one English sentence can be translated into many Myanmar sentences. Therefore, we prepared two forms of translation style in our MP corpora such as one-to-one and many translations.

A. One-to-One Translation

In this translation form, one English sentence is translated into one meaningful, polite and written form of Myanmar sentence as follows:

English: My name is Mg Mg.
Myanmar: ကျွန်ုပ် ၏ အမည် သည် မောင်မောင် ဖြစ် ပါသည် ။
English: I like this cake.
Myanmar: ကျွန်ုပ် သည် ဤ ကိတ်မုန့် အား ကြိုက်နှစ်သက် ပါသည် ။

B. One-to-Many Translation

Unlike the above translation form, one English sentence is translated into many possible Myanmar sentences because there are many types of Myanmar pronouns, daily conversation style and sentence ending words.

For example, "I" can be translated into "ကျွန်ုပ် (I)", "ကျွန်တော် (I)", "ကျနော် (I)", "ကျွန်မ (I)", "ကျမ (I)", "ကျုပ် (I)", "ငါ (I)" and "is" can be "ရှိပါသည် (is)", "ရှိသည် (is)", "ရှိပါတယ် (is)", "ရှိတယ် (is)", "ရှိတယ်လေ (is)", "ဖြစ်ပါသည် (is)", "ဖြစ်သည် (is)", "ဖြစ်ပါတယ် (is)", "ဖြစ်တယ် (is)", "ဖြစ်တယ်လေ (is)" in Myanmar.

For example, we can translate the "My name is Mg Mg." English sentence into several Myanmar sentences as shown below.

English: My name is Mg Mg.
Myanmar: ကျွန်ုပ် ၏ အမည် သည် မောင်မောင် ဖြစ် သည် ။

ကျွန်ုပ် ရဲ့ နာမည် က မောင်မောင် ဖြစ် ပါတယ် ။
ကျွန်ုပ် ရဲ့ နာမည် က မောင်မောင် ဖြစ် တယ် ။
ကျွန်ုပ် ရဲ့ နာမည် က မောင်မောင် ပါ။
ကျွန်ုပ် ရဲ့ နာမည် က မောင်မောင် လေ ။
ကျွန်တော့် ရဲ့ နာမည် က မောင်မောင် ဖြစ် ပါတယ် ။
ကျွန်တော့် နာမည် က မောင်မောင် လေ ။
ကျုပ် နာမည် က မောင်မောင် ပါ။
ငါ့ နာမည် က မောင်မောင် လေ ။

In this study, we prepared MP one-to-many corpus that contained 8,394 English sentences and 46,758 translated Myanmar sentences. Some English sentences are translated into 10 or 20 or 30 Myanmar sentences and so on. The maximum number of translations from one English sentence into Myanmar is 1,448.

V. PHRASE-BASED STATISTICAL MACHINE TRANSLATION(PBSMT)

A PBSMT translation model is based on phrasal units [11]. Here, a phrase means a contiguous sequence of words and generally, not a linguistically motivated phrase. Typically, phrase-based translation model gives better translation performance than word-based models. We can describe a simple phrase-based translation model consisting of phrase-pair probabilities extracted from corpus and a basic reordering model, and an algorithm to extract the phrases to build a phrase-table [12].

The phrase translation model is based on noisy channel model. To find best translation $e^{\wedge}$ that maximizes the translation probability P(e|f) given the source sentences; mathematically. Here, the source language is French and the target language is English. The translation of a French sentence into an English sentence is modeled as (1).

$$e^{\wedge} = argmax_e P(e|f) \quad (1)$$

Applying the Bayes' rule, we can factorized the P(e|f) into three parts as (2).

$$P(e|f) = \frac{P(e)}{P(f)} P(f|e) \quad (2)$$

The final mathematical formulation of phrase-based model is as (3).

$$\text{argmax}_e P(e|f) = \text{argmax}_e P(f|e)\, P(e) \quad (3)$$

We note that denominator P(f) can be dropped because for all translations the probability of the source sentence remains the same. The P(e|f) variable can be viewed as the bilingual dictionary with probabilities attached to each entry to the dictionary (phrase table). The P(e) variable governs the grammatically of the translation and we model it using n-gram language model under the PBSMT paradigm.

VI. EXPERIMENTAL SETUP

A. Corpus Statistics

For experiments, both one-to-one and many translations corpora contain sentences from Myanmar primary level English textbooks. Myanmar3 Unicode font is used for both Myanmar and English sentences. In one-to-one translation corpus, there are a total of 8,394 parallel sentences: 6,716 (80% of total sentences) sentences for training, 839 (10% of the remaining total sentences) sentences for development and evaluation. And, there are a total of 46,758 parallel sentences in one-to-many translations corpus. As presented in section IV, we considered the fact that one English sentence can be translated into several Myanmar sentences. This makes us more motivation to study the performance of PBSMT with respect to the various Myanmar translated sentences.

Therefore, we prepared different datasets for several experiments. For example, in the dataset for experiment-1, each English sentence is translated into the maximum 10 Myanmar sentences. Similarly, in the dataset for experiment-2, each English sentence is translated into the maximum 20 Myanmar sentences and so on. Therefore, the dataset for final experiment contained the maximum 1,448 Myanmar translated sentences for each English sentence. And, the different datasets are divided into training (80%), development (10%), and testing (10%) datasets respectively. There are no overlap of parallel sentences between training, development, and testing datasets.

B. Word Segmentation

A core issue in SMT is the identification of translation units. In phrase-based SMT, these units are comprised of bilingual pairs consisting of sequences of source and target tokens (words). Therefore, word segmentation (which defines the nature of these tokens) is one of the key preprocessing steps in SMT [13]. In this paper, we collect the necessary data as presented in section IV. As we know, Myanmar sentences are written as contiguous sequences of syllables and they are usually not separated by white space. Spaces are used

for separating phrases for easier reading. However, it is not strictly necessary, and these spaces are rarely used in short sentences. There are no clear rules for using spaces in Myanmar language, and thus spaces may (or may not) be inserted between words, phrases, and even between a root words and their affixes [13].

In this study, we did manual segmentation process to identify the word boundary by using five rules which are applied by proposed myPOS. These five rules are described with some examples as follows [14]:

- Myanmar word can usually be identified by the combination of root word, prefix and suffix.
  Unsegmented Word: သွားသည်
  Segmented Word: သွား သည်
- Plural Nouns are identified by following the particle.
  Unsegmented Word: ကျောင်းသားများ
  Segmented Word: ကျောင်းသား များ
- Possessive words are identified by following post positional marker.
  Unsegmented Word: သူမ၏ အဖေ
  Segmented Word: သူမ ၏ အဖေ
- Noun is identified with the combination of particle to the verb or the adjective.
  Unsegmented Word: ကျန်းမာရေး၊ ခင်မင်မှု
  Segmented Word: ကျန်းမာ ရေး၊ ခင်မင် မှု
- Particle state the type of noun, and used after number or text number.
  Unsegmented Word: စာအုပ်၂အုပ်၊ ပန်းသီးငါးလုံး
  Segmented Word: စာအုပ် ၂ အုပ်၊ ပန်းသီး ငါး လုံး

Besides, in our manual word segmentation rules, compound nouns are considered as one word and thus, a Myanmar compound word "လက်ဖက်ရည် + အိုး" ("tea" + "pot" in English) is segmented as one word "လက်ဖက်ရည်အိုး". Myanmar adverb words such as "စောစောစီးစီး" ("early" in English) are also considered as one word.

C. Moses SMT System

We used the PBSMT system provided by the Moses toolkit [15] for training the PBSMT statistical machine translation systems. The word segmented source language was aligned with the word segmented target language using GIZA++ [16]. The alignment was symmetrized by grow-diag-final and heuristic [11]. The lexicalized reordering model was trained with the msd-bidirectional-fe option [17]. We used KenLM [18] for training the 5-gram language model with modified Kneser-Ney discounting [19]. Minimum error rate training (MERT) [20] was used to tune the decoder parameters and the decoding was done using the Moses decoder (version 2.1.1) [15]. We used default settings of Moses for all experiments.

D. Evaluation

Two automatic criteria are used for the evaluation of the machine translation output. One was the de facto standard automatic evaluation metric Bilingual Evaluation Understudy (BLEU) [21] and the other was the Rank-based Intuitive Bilingual Evaluation Measure (RIBES) [22]. The BLEU score measures the precision of n-gram (over all $n \leq 4$ in our case) with respect to a reference translation with a penalty for short translations [21]. Intuitively, the BLEU score measures the adequacy of the translation and large BLEU scores are better. RIBES is an automatic evaluation metric based on rank correlation coefficients modified with precision and special care is paid to word order of the translation results. The RIBES score is suitable for distance language pairs such as Myanmar and English. Large RIBES scores are better.

VII. RESULT AND DISCUSSION

The BLEU and RIBES score results for machine translation experiments with PBSMT between Myanmar and English languages are shown in Table I and II. Here, bold numbers indicate the highest scores among several PBSMT experiments. The RIBES scores are shown in the round brackets. "My" stands for Myanmar, "En" stands for English respectively.

In one-to-one MT, English-Myanmar translation achieved 59.28 BLEU and 0.8468 RIBES scores and Myanmar-English translation achieved 89.42 and 0.9077 RIBES scores using PBSMT approach.

When we measured the performance of PBSMT using one-to-many translation corpora, we found that the BLEU and RIBES scores are gradually increased in both English to Myanmar and Myanmar to English translations as shown in Table II. We carried out these machine translation experiments by incrementing the number of translated Myanmar sentences. In other words, each English sentence is translated into 10, 20, 30, ..., 100, 200, 300, …, 1,448 translated Myanmar sentences.

From the English to Myanmar translation results with one-to-many corpora (see Table II), it can be seen clearly that the gain in BLEU and RIBES scores of 1-20 translation model significantly increased than 1-10 translation model (from 19.68 to 46.28 in terms of BLEU score and from 0.6969 to 0.8118 in terms of RIBES score). From the models 1-30 to 1-90 and 1-200 to 1-1100 translation results, we can assume that increasing the number of translated Myanmar sentences slightly impact (only a small fraction) on PBSMT performance. On the other hand, the results of the 1-100 and 1-1200 models gains significant BLEU scores (+1.64 and -1.52 in average). One more factor we should consider is 839 sentences of the test-set that we used for all one-to-many corpora experiments.

According to the results of Myanmar to English translation models, we found that the gains in BLEU score continuously increased (50.17 ~ 71.15 BLEU) between 1-10 and 1-400 translation model. However, this 71.15 result slightly decreased (average 0.25 BLEU) in 1-500 model. Then, there were alternate changes in terms of BLEU and RIBES scores. The gain in BLEU score of 1-800 translation model is much larger (average 2.18 BLEU) than 1-700 translation model. Generally, English to Myanmar translation results are above 50 BLEU scores for 1-700 to 1-1448 models and the highest BLEU and RIBES scores are achieved by 1-1448 (52.38 BLEU score) and 1-1300 (0.8238 RIBES score) translation models. Similarly, Myanmar to English translation results are above 70 BLEU scores from 1-400 to 1-1448 models and the highest BLEU and RIBES scores (75.12 and 0.8838) are achieved by 1-1300 translation model. Our results with current one-to-one test dataset indicate that Myanmar to English machine translation is better performance (around 23 BLEU and 0.06 RIBES scores higher) than English to Myanmar translation direction.

TABLE I. BLEU AND RIBES SCORES OF PBSMT FOR ONE-TO-ONE TRANSLATION CORPUS BETWEEN MYANMAR AND ENGLISH

Corpus Size	En-My	My-En
training = 6716 development = 839 testing = 839	59.28 (0.8468)	89.42 (0.9077)

TABLE II. BLEU AND RIBES SCORES OF PBSMT FOR ONE-TO-MANY TRANSLATION CORPUS BETWEEN MYANMAR AND ENGLISH

No. of Myanmar Translated Sentences (En-My) Corpus Size [training, dev, test]	En-My	My-En	No. of Myanmar Translated Sentences (En-My) Corpus Size [training, dev, test]	En-My	My-En
1-10 [3591, 449, 449]	19.68 (0.6969)	50.17 (0.8245)	1-400 [22937, 2866, 2867]	49.71 (0.8209)	71.15 (0.8779)
1-20 [6081, 760, 760]	46.28 (0.8118)	57.05 (0.8439)	1-500 [24813, 3102, 3102]	49.80 (0.8174)	70.90 (0.8746)
1-30 [7878, 985, 985]	46.13 (0.8052)	61.52 (0.8639)	1-600 [25417, 3177, 3177]	49.62 (0.8181)	71.79 (0.8790)
1-40 [9255, 1157, 1157]	46.94 (0.81061)	61.97 (0.8650)	1-700 [25831, 3229, 3229]	50.33 (0.8220)	72.79 (0.8834)
1-50 [10313, 1289, 1289]	47.58 (0.8105)	63.22 (0.8670)	1-800 [26269, 3284, 3284]	50.74 (0.8192)	72.38 (0.8812)
1-60 [11098, 1387, 1387]	46.82 (0.8158)	64.48 (0.8691)	1-900 [26718, 3839, 3839]	51.10 (0.8224)	74.56 (0.8830)
1-70 [11777, 1472, 1472]	46.28 (0.8129)	66.86 (0.8758)	1-1000 [27114, 3388, 3389]	50.79 (0.820577)	74.08 (0.878125)
1-80 [12398, 1550, 1550]	46.96 (0.8110)	67.54 (0.8753)	1-1100 [27297, 3412, 3412]	51.85 (0.8225)	72.95 (0.8827)

1-90 [12951, 1619, 1619]	46.78 (0.8115)	67.54 (0.8773)	1-1200 [27434, 3429, 3429]	50.33 (0.8213)	72.87 (0.8799)
1-100 [13470, 1684, 1684]	48.42 (0.8179)	68.54 (0.8778)	1-1300 [27585, 3448, 3448]	51.13 **(0.8238)**	**75.12** **(0.8838)**
1-200 [17698, 2212, 2212]	49.11 (0.8117)	69.43 (0.8758)	1-1400 [27704, 3462, 3462]	52.09 (0.8224)	74.10 (0.8749)
1-300 [20789, 2599, 2599]	50.07 (0.8216)	69.76 (0.8752)	1-1448 [27741, 3468, 3468]	**52.38** (0.8173)	74.49 (0.8777)

From the overall results of Table I and II, both one-to-one and one-to-many models shown that Myanmar to English machine translation achieved better performance than English to Myanmar translation direction. Here, note on corpus size differences (including development and test datasets) among one-to-many models (see Table II). Although, we cannot directly compare between one-to-one and one-to-many model results, we found that the best BLEU and RIBES scores of one-to-many are lower than one-to-one for both My-En and En-My translation directions (BLEU: 52.38 < 59.28, RIBES: 0.8173 < 0.8468 for En-My and BLEU: 74.12 < 89.42, RIBES: 0.8838 < 0.9077 for My-En).

However, the series of BLEU and RIBES scores of the one-to-many models (see Table II) proved that multiple translations of English to Myanmar gradually increased the machine translation performance for both En-My and My-En.

VIII. ERROR ANALYSIS

For both one-to-one and many translation corpora, we analyzed the translated outputs using Word Error Rate (WER) [23]. We also used the SCLITE (score speech recognition system output) program from the NIST scoring toolkit SCTK version 2.4.10 [24] for making dynamic programming based alignments between reference (ref) and hypothesis (hyp) and calculation of WER. The formula for WER can be stated as (4):

$$\text{WER} = \frac{S+D+I}{N} = \frac{S+D+I}{S+D+C} \quad (4)$$

where S is the number of substitutions, D is the number of deletions, I is the number of insertions, C is the number of correct words and N is the number of words in the reference ($N = S + D + C$) [23]. It is needed to note that if the number of insertions is very high, the WER can be greater than 100%.

The following examples show WER calculation on the translated outputs of PBSMT approach for Myanmar-English language pair with two types of corpora. The first one is WER calculation for the use of one-to-one Myanmar translation corpus. For example, scoring I, D and S for the translated Myanmar sentence "ကျနော် ကြိုက် သော အစားအစာ က ပေါင်မုန့် ဖြစ် ပါ တယ် ။" ("My favourite food is bread ." in English) compare to a reference sentence, the output of the SCLITE program is as follows:

Scores: (#C #S #D #I) 9 1 1 0
REF: ငါ့ ရဲ့ ကြိုက် သော အစားအစာ က ပေါင်မုန့် ဖြစ် ပါ တယ် ။
HYP: ********* ကျနော် ကြိုက် သော အစားအစာ က ပေါင်မုန့် ဖြစ် ပါ တယ် ။
Eval: D S

In this case, one substitution (*** ==> ငါ့) and one deletion (ရဲ့ ==> ကျနော်) happened, that is C = 9, S = 1, D = 1, I = 0, N = 11 and and thus its WER is equal to 18%. The following is for Myanmar-English translation example and all translated words are correct, C = 6, S = 0, D = 0, I = 0, N = 0 and its WER is equal to 0%.

Scores: (#C #S #D #I) 6 0 0 0
REF: my favourite food is bread .
HYP: my favourite food is bread .
Eval:

The next one is WER calculation for English-Myanmar with one-to-many Myanmar translation corpus. For example, scoring I, D and S for the translated Myanmar sentence "ဒါက ငါ့ အဘိုး ဖြစ် ပါတယ် ။" ("This is my grandfather.") in English) compare to a reference sentence, the output of the SCLITE program is as follows:

Scores: (#C #S #D #I) 7 1 0 1
REF: ဒါက ငါ့ *** အဘိုး ဖြစ် ပါတယ် ။
HYP: ဒါက ငါ့ ရဲ့ *** GRANDFATHER ဖြစ်ပါတယ်။
Eval: I S

In this case, one substitution (*** ==> ရဲ့) and one insertion (အဘိုး ==> GRANDFATHER) happened, that is C = 7, S = 1, D = 0, I = 1, N = 8 and thus its WER is equal to 25%. The following is for Myanmar-English translation example. In this case, one substitution အဘိုး ==> GRANDFATHER) happened, that is C = 4, S = 1, D = 0, I = 0, N = 5 and thus its WER is equal to 20%.

Scores: (#C #S #D #I) 4 1 0 0
REF: this is my GRANDFATHER .
HYP: this is my အဘိုး .
Eval: S

Fig. 1 and 2 present the average WER percentages of one-to-one and one-to-many translation models. The results show that "Myanmar-English" translation gave the lower WER value than "English-Myanmar" translation.

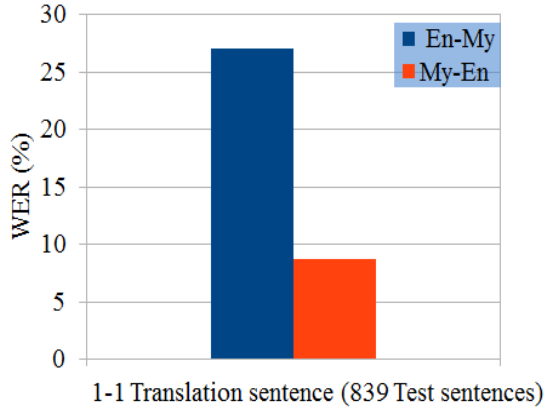

Figure 1. Average WER% for PBSMT, with one-to- one translation corpus (lower is better)

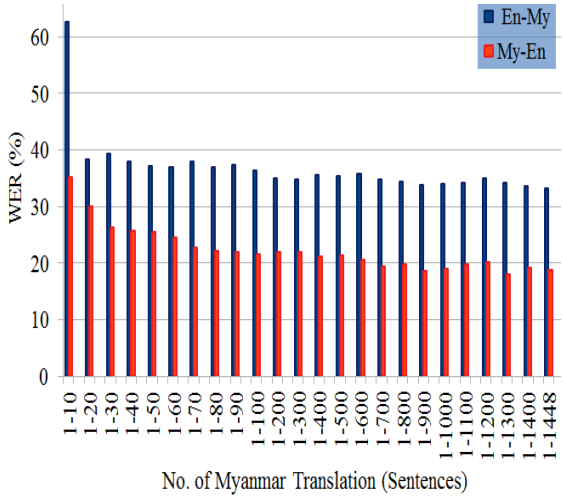

Figure 2. Average WER% for PBSMT, with one-to- many translation corpus (lower is better)

After we made analysis of the confusion pairs of PBSMT model in details, we found that most of the confusion pairs are caused by (1) the nature of Myanmar languages (written or speaking form), (2) unknown short form (3) ambiguous article mistakes and (4) limited size of the training data especially on English language. For example, the top 10 confusion pairs of one-to-many translation corpus based PBSMT translation model are shown in Table III. In this table, the 1st column is the reference and hypothesis pair (i.e. output of the PBSMT translation model) for English to Myanmar translation. The third one is for that of Myanmar to English translation.

All of the confusion pairs in 1st column are caused by the nature of Myanmar language. For example, in Myanmar written or speaking form, the words "ဟုတ် ("is" in English)" are the same with the words "ဖြစ် ("is" in English)". Also, the words "က ("is" in English)" and "သည် ("is" in English)" in the subject place and the words "ရဲ့ ("of or 's" in English)" and "၏("of or 's" in English)" in the possessive place are the same meanings. In other words, these hypotheses are synonyms of the reference words.

TABLE III. THE TOP 10 CONFUSION PAIRS OF PBSMT MODEL USING ONE-TO-MANY TRANSLATION CORPUS BETWEEN MYANMAR AND ENGLISH

En-My (Ref ➔ Hyp)	Freq	My-En (Ref ➔ Hyp)	Freq
ရဲ့ ➔ ၏	29	i'm ➔ am	14
ဟုတ် ➔ ဖြစ်	25	uncle ➔ ဦးေလး	8
ဒီဟာ ➔ ဒါ	24	window ➔door	8
နေ ➔ ဖြစ်	24	pudding ➔ပူတင်း	7
အဲ့ဒါ ➔ အဲဒါ	22	am ➔ the	6
ဒါ ➔ ဒီဟာ	21	it's ➔ is	6
ဒီဟာ ➔ က	17	sing ➔ သီချင်းဆို	6
က ➔ သည်	15	a ➔ an	5
တဲ့ ➔ သော	14	an ➔ a	5
ကျနော် ➔ ရဲ့	13	aunt ➔ အဒေါ်	4

Also, for the machine translation from Myanmar to English, the confusion pairs of "i'm ➔ am" and "it's ➔ is" are caused due to unknown short form. And, we found that the confusion pairs of "am ➔ the", "a ➔ an" and "an ➔ a" are caused by the

ambiguous article mistakes. And, the confusion pairs of "window ➔ door" and "pudding ➔ ပုတ်ငုံး" and so on are related to the limited size of our training data. Thus, the translation models couldn't learn well.

IX. CONCLUSION

This paper contributes the first PBSMT machine translation evaluation between Myanmar and English languages for specific primary educational domain in Myanmar. We used over 8K Myanmar-English parallel sentences as one-to-one translation corpus and over 46K parallel sentences as one-to-many translation corpus. We analyzed the performance differences of PBSMT translation models by using several number of Myanmar translated sentences (1 English sentence to 10 or 20 of 30 Myanmar translated sentences and so on). The results proved that the highest BLEU and RIBES scores (52.38 and 0.8238 for English-Myanmar and 75.12 and 0.8838 for Myanmar-English) can be achieved for Myanmar-English language pair with one-to-many translation corpus. This paper also presents detail analysis on confusion pairs of machine translation between Myanmar-English and English-Myanmar. In the near future, we plan to extend our experiments with other SMT approaches such as Hierarchical Phrase Based Statistical Machine Translation (HPBSMT) and Operation Sequence Model (OSM) on the one-to-many parallel corpus.

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Generating Myanmar News Headlines using Recursive Neural Network

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Abstract

Text summarization in the form of Headline prediction for written articles becomes a popular research recently. This paper presents a headline prediction model using Recursive Recurrent Neural Network (Recursive RNN) for Myanmar articles and evaluates the performance by comparing with sequence-to-sequence models. Recursive RNN model use encoder-decoder LSTM model that generates a single word forecast and call it recursively to text summarization in the form of news headlines for Myanmar news articles. A Recursive RNN, which takes input as the sequence of variable length and has to generate the variable length output by taking into account the previous input. 5000 articles of Myanmar news were collected to train headline prediction model. The performance of Recursive RNN, Seq2Seq with one-hot encoding and Seq2Seq with word embedding (GloVe) were evaluated in terms of ROUGE score values. The experimental results show that Recursive RNN model significantly better than other two models.

Keywords—*LSTM, Recursive RNN, ROUGE, word embedding*

I. INTRODUCTION

Huge amount of information are available on the internet news websites but users want to extract best particular information within a short search time. Document summarization has two types extractive and abstractive. Extractive method chooses the salience sentences from the original document. Abstractive method uses the encoder-decoder architecture which generates latent factor representation of the data and decodes it to generate a summary1. Abstractive model can be more effective by performing generation from scratch but slow and inaccurate encoding of very long documents [9]. Recursive RNN-based text summarization model was popular generating a new headline or short summary consisting of a few sentences that extract the most important point of article or a passage. Recursive RNN based summarization model to be very effective for transforming Unicode Myanmar text from one to another. These models are trained on large amount of input sequence of Myanmar words and expected output sequence of Myanmar words and generate output sequences based on the given input. Text summarization is the most challenging problem in Myanmar Language. There are many automatic text summarizers for other languages but a few for Myanmar language. Therefore, as our motivation, we was collected Myanmar news corpus including title of Myanmar news articles and description of text using Myanmar standard Unicode font in Myanmar news web pages. Three contributions are described in this paper. The first one is obtain the context vectors from the Myanmar news document which can be used for extractive summarization task. The second one is using RNN headline training model for Myanmar news summarization. The third one is using document context vector, produce the Myanmar news summarization. The rest of the paper is organized as follows: section 2 describes related work and section3 describes challenges of Myanmar news summarization section4 describe the overview of RNN-based Myanmar text summarization. Section 5 experimental setups and evaluation results and section 6 discusses conclusion and future work direction.

[1] https://mc.ai/extractive-and-abstractive-text-summarization-methods/

II. RELATED WORK

The author proposed abstractive text summarization using Attention Encoder-Decoder Recurrent Neural Networks and modeling key-words. In this paper contain capturing the hierarchy of sentence-to-word structure, and producing rare word or unseen word at training time [1]. Khatri, Chandra, Gyanit Singh, and Nish Parikh proposed Document-context Sequence to sequence (Seq2Seq) learning has recently been used for abstractive and extractive summarization. It gives humans high-level understanding of the document and Seq2Seq model will generate much richer document specific summaries [2]. In this paper, using headline generation models, train the English GigWord corpus (Graff & Cieri, 2003), consisting of news articles from number of publishers. Summarize the first sentence of an article is used to predict the headline, using RNN model, RNN-based encoder-decoder models with attention (seq2seq) perform well ROUGE (Lin, 2004), an automatic metric often used in summarization, and human evaluation [3]. Present Recursive Neural Networks (R2N2) for the sentence ranking task of multi-document summarization. It transforms the ranking task into a hierarchical reversion process which is modeled by recursive neural networks [4]. In this paper describe about the performance of LSTM bidirectional and Seq2Seq model on Amazon reviews and news dataset is compared using BLEU, ROUGE_1 and ROUGE_2 score [5]. In the paper [6] describe about a fully data-driven approach to abstractive sentence summarization. This technique useful for local attention-based model that generates each word of the summary based on the source sentence. Although the model is simple, it can be easily trained on a large amount of training dataset. This paper emphasized on feed-forward neural network with attention-based encoder to resolve the challenge of abstractive summarization. Briefly explain attentive recurrent neural network and recurrent neural network encoder-decoder. Amazon reviews dataset and Bidirectional LSTM model using Sequence to Sequence model compared with the human generated summary [7].

III. NATURE AND COLLATION OF MYANMAR LANGUAGE AND SEGMENTATION

The nature of Myanmar script has 33 consonants and the consonants combine with vowels and medial to form the complete syllable in Myanmar language. Myanmar language does not place spaces between words but spaces are usually used to separate phrases. In Myanmar language, sentences are clearly delimited by a sentence boundary marker pote ma (။) but words are not always delimited by spaces. In Myanmar language segmentation process is the fundamental and very important step in generating the summaries. Segmentation process includes sentence and word segmentation that are finding boundaries of sentences and word tokens. In English, word boundaries can be easily defined but it is not easy in Myanmar. Myanmar words are written continuously without using space in sentences. Therefore word segmentation process is very important phase to pass word tokens and sentences to next steps of summarization. For Myanmar sentence and word segmentation, we used Myanmar word segmenter of UCSY [8]. For example,

Input Myanmar Sentence:
ပြင်ဦးလွင်မြို့ဒီဇင်ဘာခြုံတွင်ပန်းအလှဆင်ပွဲတော်ကျင်းပမည်။
Segmented Sentence:
ပြင်ဦးလွင်မြို့_ ဒီဇင်ဘာ_ ခြုံ_ တွင်_ ပန်း_ အလှ_ ဆင်_ ပွဲတော်_ ကျင်းပမည်_ ။

The overview of the proposed system described in figure 1. The proposed system used the concept of machine learning based Recursive RNN predicted headline summary. And then preprocessing steps are performed. In preprocessing step, the system performs word segmentation, stop words removal such as pronouns, prepositions, conjunction and particles etc.

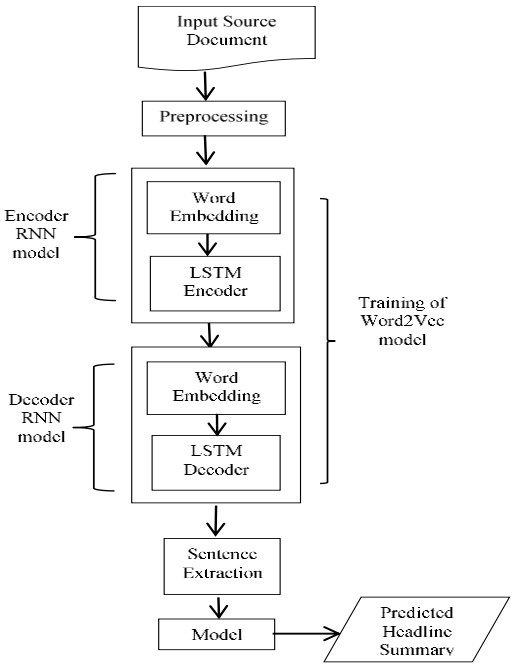

Figure 1. Recursive RNN-based Summarization model overview

And then Headline Generator is using the Recursive RNN-based model and consists of encoder and decoder model, the encoder use one RNN and the decoder use another RNN. The encoder encodes the sequence of text and builds the context vector. Secondly, the decoder read the input sequence from the encoder and generates the predicted output sequence of headline summary.

IV. EXPERIMENTAL SETUP

In this section, explained about experiments that conduct better understands neural headline generation system and describe about data preparation and data processing.

A. Data Preparation

Myanmar news articles are collected written with Myanmar Unicode font from Myanmar News websites. In this experiment, we used a collection of web pages from BBC Burmese news[2], Irrawaddy Burmese news[3], Weekly Eleven Burmese news[4], Mizzima Burmese news[5]. To build effective headline prediction models, we had to prepare data preparation, data cleaning and the model architecture. Preprocessed data take as the input to the neural network for training and validation. In this data meaningful sentences are included.

TABLE I. DATA STATISTIC OF MYANMAR NEWS DATASET

Data	Article	Sentence
Training Sentences	3211	32110
Validation Sentences	412	4120
Testing Sentences	1377	13770

B. Data Processing

In my training corpus have two column, title and description of text. It can be save CSV format. The titles were used as training target. Before the raw text can be fed into the neural network, it has to be preprocessed. In preprocessing stage sentence segmentation and word segmentation are correctly processed. The input texts are split into an article and summary part. We used preprocessed article text take as input X and the preprocessed summary is target product Y. In data preprocessing includes extracting contents and titles. In this study, training model use the training data 37.7 M news articles with 898 M words. We collected Myanmar news from various Myanmar News websites. The collected news is converted into CSV format and takes as input to the neural architecture for training and validation. Load the CSV input file using python Pandas library [12]. Pandas provide a platform for handling the data in a data frame. It contains many open-source data analysis tool written in Python, the methods to check missing data merge data frames and reshape data structure etc. And then split the dataset into input and output variables for machine learning and apply preprocessing data transforms to the input variables and then finally produce the summarized data as outputs. In training time, we use dimension words embedding by the word2vec algorithm and use epoch for training time. We have used randomly shuffled the training data at every epoch. Training model is developed to identify the sub sequence of words or text that needs to summarize and consider the input text as a sequence of words or text. In extraction phases, summarization model extract the key sentences from the Myanmar news. We implemented our model on the seq2seq and encoder-decoder recurrent networks in Keras library. Keras provides suitable libraries to load the dataset and split it into training set and test set [10]. RNN based Encoder-Decoder model using Google deep learning framework, Tensorflow. Encoder encodes the input Myanmar articles news as the context vector. Decoder reads the encoded input text sequence of text from the encoder and generates the output text sequences.

V. METHOLOGY

A. Neural Headline Generation Model

Neural headline generation targets at learning a model to map a short text into a headline. Neural headline generation means to map documents to headlines with recurrent neural network. Given an input document $X=(x_1,...,x_m)$ where each word x_i get from a fixed vocabulary V, the neural headline generation model to take X as input, and generates a short headline $Y=(y_1,\ldots,y_n)$ with length $n<m$ word by word. Headline generator consists of an encoder and decoder which are constructed with Long Short Term

[2] https://www.bbc.com/burmese
[3] https://burma.irrawaddy.com/category/news
[4] https://news-eleven.com/news
[5] http://www.mizzimaburmese.com/news
6https://en.wikipedia.org/wiki/Recursive_neural_network

Memory (LSTM), which is one of the recurrent neural networks. The encoder constructs distributed representation from the title sentence of Myanmar article news and decoder generated Myanmar headlines news from the distributed representation.

B. Recursive Recurrent Neural Network

Recursive Recurrent Neural Network (Recursive RNN) is the deep neural network by applying the same set of weight recursively to produce a variable size input structure. Recursive RNN using parse-tree based structure representation.6 Recursive RNN recursively merge pairs of word and phrase representation. Recursive RNN based encoder-decoder neural network architecture. The encoder takes the input text of a news articles one word at a time. Each word is passed through an embedding layer that transforms the words into distributed word representation. The distributed representations of words are combined using multi-layer neural network with the hidden layers generated after serving in the previous word. The decoder must generate two type of information. The first one is context vector (or) fixed length vector and the second one is generated output sequence as a predicted summary. The encoder-decoder architecture means recurrent neural networks for sequence prediction problems that have a variable number of inputs text, outputs text or both inputs and outputs text [11]. The encoder reads the entire input sequence of variable length of text and encodes it into an internal representation of words called the context vector. The decoder reads the encoded the variable input sequence of length from the encoder and generates the output sequence of the predicted headline. In figure 2 shows the encoder and decoder architecture and text is input and headline is generated as output.

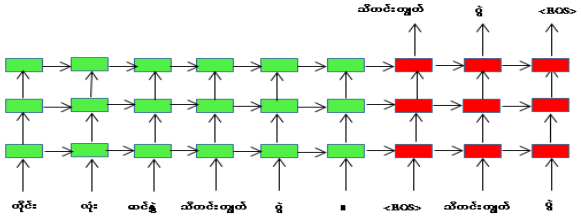

Figure 2. Example Encoder-Decoder neural network architecture

C. Baseline Sequence-to-Sequence with one hot encoding

To verify the effectiveness of the proposed method, has investigated various approaches for comparison. In this work, as the baseline model has considered Sequence to Sequence with one-hot encoding summarization model and Sequence to Sequence with word embedding (GloVe) summarization model. In sequence to sequence one hot encoding model, the sequence provides at least one example of every possible value in the sequence. It uses automatic methods to define the mapping of labels to integers and integers to binary vectors7.

D. Baseline Sequence-to-Sequence with word embedding (GloVe)

Sequence-to-sequence with word embedding (GloVe) summarizer use for encoder input. GloVe means word embedding because the global corpus statistic is captured directly by the model. Global model are widely used for learning distributed word representation. This model is unsupervised learning algorithm for obtaining vector representation for each word GloVe model, as well as some auxiliary tools to construct word-word co-occurrence matrices from large corpora8. The model is an unsupervised learning algorithm for obtaining vector representations for words [14].

VI. Evaluation

The performance of the model was measured by two different ways. The first way is ROUGE score evaluation and the second way is training and hold out loss. ROUGE (Recall-Oriented Understudy for Gisting Evaluation method based on N-gram statistics, ROUGE-1, ROUGE-2 and ROUGE-L, ROUGE-SU which are computed using the matches of unigrams, bigrams and longest common subsequences and skip-bigram plus unigram-based co-occurrence statistics respectively [13].

[7] https://machinelearningmastery.com/how-to-one-hot-encode-sequence-data-in-python/
[8] https://nlp.stanford.edu/projects/glove/

A. Experimental Results

Three models are considered as our assessments evaluation. These models produce the results of original headlines and predicted headlines. In table 2, table 3 and table 4 shows the ROUGE-1, ROUGE-2 and ROUGE-L evaluation results of three models. From the experimental results of Table 2, Table 3 and Table 4, scores are given for ROUGE-1, ROUGE-2 and ROUGE-L, word overlap between a systems generated summary and reference summary. We have noticed that Recursive RNN based summarizer gives the better results of ROUGE-1, ROUGE-2 and ROUGE-L than other two models.

TABLE II. ROUGE-1 SCORE RESULTS FOR THREE MODELS

ROUGE-1			
Model	***Precision***	***Recall***	***F-score***
Seq2Seq with one-hot encoding	0.65377	0.53109	0.56808
Seq2Seq with word embedding	0.36879	0.30019	0.33097
Recursive RNN	**0.71322**	**0.70159**	**0.70736**

TABLE III. ROUGE-2 SCORE RESULTS FOR THREE MODELS

ROUGE-2			
Model	***Precision***	***Recall***	***F-score***
Seq2Seq with one-hot encoding	0.28497	0.22821	0.25345
Seq2Seq with word embedding	0.11774	0.08671	0.09609
Recursive RNN	**0.37352**	**0.36706**	**0.37027**

TABLE IV. ROUGE-L SCORE RESULTS FOR THREE MODELS

ROUGE-L			
Model	***Precision***	***Recall***	***F-score***
Seq2Seq with one-hot encoding	0.12234	0.13241	0.23385
Seq2Seq with word embedding	0.09052	0.09987	0.14902
Recursive RNN	**0.11222**	**0.18937**	**0.31844**

Table 5 shows the example of generated headlines for articles from various Myanmar news websites.

In figure 3, figure 4 and figure 5 shows the training plot below was obtain by running for 100 epochs by using Recursive RNN model, Sequence to Sequence with one hot encoding model and seq2seq with word embedding model. Line plot shows the y-axis and x-axis. Y-axis shows the accuracy of the model and x-axis shows the number of epoch. In figure 3 shows the evaluation metrics as a function of training epoch and we can see that the train accuracy values are relatively higher than the validation accuracy. In figure 3 shows the model in which training loss suddenly decreases but validation loss eventually goes up.

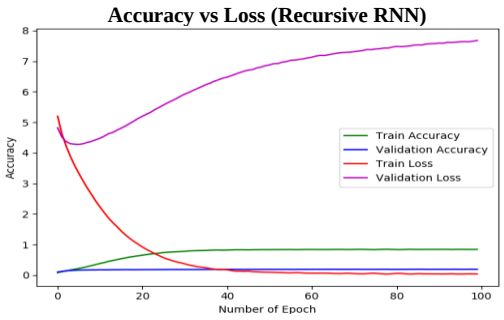

Figure 3. accuracy vs loss with Recursive RNN

In Fig. 4 shows the learning curve of Seq2Seq with one hot encoding model. From the output, we can see that the train accuracy slightly higher than the validation accuracy.

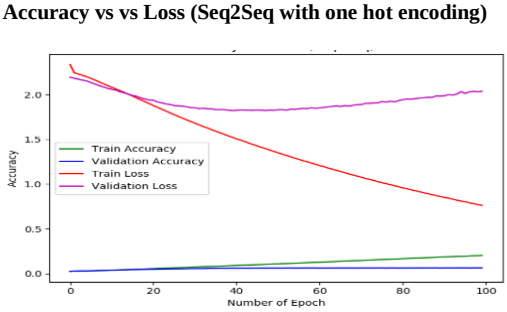

Figure 4. accuracy vs loss with seq2seq with one hot encoding

In fig 5 shows the learning curve of Seq2Seq with word embedding (GloVe) model. The output result shows that difference between train accuracy and validation accuracy. According to the training results, Recursive RNN-based headline summarization model is good performance results than other two models.

TABLE V. EXAMPLE OF MYANMAR NEWS PREDICTED HEADLINE

Text	Actual Headline	Predicted Headline
အောင်လအန်ဆန် ၏ ပွဲစဉ် အား ကြည့်ရှုအားပေး စဉ် အသက် ၇၀ ကျော် အဘိုးအို တစ်ဦး လဲ ကျ သေဆုံးမှု ဒေးဒရဲ မြို့ တွင် ဖြစ်ပွား ခဲ့ ကြောင်း သိရသည် ။ ဧရာဝတီ တိုင်းဒေသကြီး ဒေးဒရဲ မြို့ မှော်ဘီ စု ကျေးရွာအုပ်စု ထ နိ ပတ် ကျေးရွာ တွင် အောက်တိုဘာ ၁၃ ရက် ညပိုင်း တွင် “ မြန်မာ့ ဂုဏ် ဆောင် အောင်လ ” ဟု အော်ဟစ် အားပေး ကာ လဲ ကျ သေဆုံး ခဲ့ ခြင်း ဖြစ်ကြောင်း ကျေးရွာ အုပ်ချုပ် ရေး မှူး ရုံး မှ သိရသည် ။	အောင်လအန်ဆန် ၏ ပွဲစဉ် အား ကြည့်ရှု အားပေး စဉ် အသက် ၇၀ ကျော် အဘိုးအို တစ်ဦး လဲ ကျ သေဆုံး မှု ဒေးဒရဲ မြို့ တွင် ဖြစ်ပွား	အောင်လအန်ဆန် ၏ ပွဲစဉ် ကြည့်ရှု အားပေး စဉ် အသက် ၇၀ ကျော် အဘိုးအို တစ်ဦး သေဆုံး မှု ဖြစ်ပွား
ဖဖ နှစ်မြောက် တရုတ် ပြည်သူ သမ္မတနိုင်ငံ ထူထောင်ခြင်း အခမ်းအနား ကို မြန်မာနိုင်ငံ ဆိုင်ရာ တရုတ် သံရုံး မှ ပြည်ထောင်စု နယ်မြေ နေပြည်တော် တွင် ပထမဆုံး အကြိမ် အဖြစ် စက်တင်ဘာ ၂၄ ရက် တွင် ကျင်းပခဲ့သည် ။ မြန်မာ အပြည်ပြည်ဆိုင်ရာ ကွန်ဗင်းရှင်း စင် တာ (MICC) ၌ ကျင်းပ သည့် အခမ်းအနား တွင် ၂၀၁၅ ခုနှစ် သည် တရုတ် – မြန်မာ သံတမန် ဆက်ဆံရေး တည်ဆောက် ခဲ့ သည့် ၆၅ နှစ် ပြည့် သည့် နှစ် လည်း ဖြစ်သည် ဟု မြန်မာနိုင်ငံ ဆိုင်ရာ တရုတ် သံအမတ်ကြီး မစ္စတာ ဟုန် လျန် က ပြောသည် ။ တရုတ် သံအမတ်ကြီး မစ္စတာ ဟုန်လျန် က ၂၀၁၁ ခုနှစ် တွင် တရုတ် – မြန်မာ နှစ်နိုင်ငံ ဘက်စုံ မဟာဗျူဟာ မိတ်ဆွေ ဆက်ဆံရေး တည်ထောင် နိုင် ခဲ့ ပြီး ၊ တရုတ် – မြန်မာ နှစ်နိုင်ငံ သည် အနီးကပ် မိတ်ဆွေ အိမ်နီးချင်း နိုင်ငံ အဖြစ် ကံကြမ္မာ အကျိုးတူ အသိုက်အဝန်း ကိုလည်း ဖြစ်ပေါ် စေ ခဲ့ သည် ဟု တရုတ် သံအမတ်ကြီး ပြောသည် ။	ဖဖ နှစ်မြောက် တရုတ် ပြည်သူ သမ္မတ နိုင်ငံ ထူထောင်ခြင်း ခမ်းအနား ကို မြန်မာနိုင်ငံ ဆိုင်ရာ တရုတ် သံရုံး မှ ပြည်ထောင်စု နယ်မြေ နေပြည်တော် တွင် ပထမဆုံး အကြိမ် အဖြစ် စက်တင်ဘာ ၂၄ ရက် တွင် ကျင်းပခဲ့	ဖဖ နှစ်မြောက် တရုတ် ပြည်သူ သမ္မတ နိုင်ငံ ထူထောင်ခြင်း အခမ်းအနား ကို မြန်မာ နိုင်ငံ ဆိုင်ရာ တရုတ် သံရုံး မှ ပထမဆုံး အကြိမ် အဖြစ် ကျင်းပခဲ့
"လာ မည့် နိုဝင်ဘာလ ၈ ရက် အထွေထွေရွေးကောက်ပွဲ အတွက် ထိုင်းနိုင်ငံ ဘန်ကောက် မြို့ နှင့် ချင်းမိုင် မြို့ တို့တွင် ကြိုတင်မဲပေး ရန် လာရောက် လျှောက်ထားသူ စုစုပေါင်း ၈၅၄ ဦး မှ ၆၀၄ ဦး သာ မဲပေး ခွင့် ရှိပြီး ကျန် ၂၅၀ မှာ မဲ စာရင်း တွင် မ ပါဝင် ကြ ဟု သိရသည် ။ ဘန်ကောက် မြို့ မြန်မာ သံရုံး က ပြီး ခဲ့ သည့် ဇူလိုင်လ နှင့် စက်တင်ဘာလ များ တွင် ကြိုတင်မဲပေး ရန် အသိပေး ခဲ့ ပြီးနောက် လာရောက် လျှောက်ထားသူ ပေါင်း ၇၁၂ ဦး ရှိရာ မှ ၅၆၁ ဦး သာ မဲပေး ခွင့် ရပြီး ၊ မဲ စာရင်း တွင် မပါ သူ ၁၅၁ ဦး ရှိသည် ဟု ထိုင်းနိုင်ငံ ဘန်ကောက် မြို့ မြန်မာ သံရုံး မှ သံအမတ်ကြီး ဦး ဝဏ္ဏဟန် က အောက်တိုဘာလ ၁၅ ရက် ၌ မဇ္ဈိမ ကို ပြောသည် ။	လာ မည့် နိုဝင်ဘာလ ၈ ရက် အထွေထွေ ရွေးကောက်ပွဲ အတွက် ထိုင်း နိုင်ငံ ဘန်ကောက် မြို့ နှင့် ချင်းမိုင် မြို့ တို့တွင် ကြိုတင်မဲပေး ရန် လာရောက် လျှောက်ထားသူ စုစုပေါင်း ၈၅၄ ဦး မှ ၆၀၄ ဦး သာ မဲပေး ခွင့် ရှိ	လာ မည့် နိုဝင်ဘာလ ၈ ရက် အထွေထွေ ရွေးကောက်ပွဲ အတွက် လာရောက် လျှောက်ထားသူ စုစုပေါင်း ၈၅၄ ဦး မှ ၆၀၄ ဦး သာ မဲပေး ခွင့် ရှိ

Accuracy vs loss (Seq2Seq with word embedding GloVe)

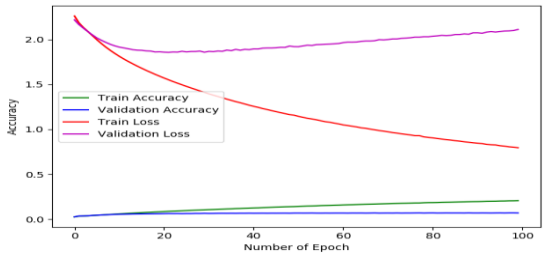

Figure 5. accuracy vs loss with Seq2Seq with word embedding (GloVe)

VII. CONCLUSION AND FUTURE WORK

We trained an encoder-decoder recurrent neural network model for generating Myanmar news headlines using the text of Myanmar news articles. The model generates a concise summary as generated outputs. Recursive RNN can be more learn complex linguistic large amount of training data. Moreover evaluations on experimental summarized results have been also discussed with related evaluation scores. According to the evaluation results, this experiment can easily provide predicted headline summaries from Myanmar news web page with acceptable evaluation scores within a particular time as the goal of our motivation. In my future work, by using larger training data sets and training these systems as auto encoders, where input is expected to match output.

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Myanmar Dialogue Act Recognition (MDAR)

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Abstract

This research aim to make the very first machine learning based Myanmar Dialog Act Recognition (MDAR) for Myanmar Dialogue System. As we know, Dialog Act (DA) recognition is the early level of dialogue understanding which can capture aspects of the user, and they are sentence-level units that represent states of a dialogue, such as greeting, question, inform, and so on. We focus on the current works about DA recognition, especially for Myanmar Dialogue. In this work, we used two machine learning approaches, which are Naïve Bayes classifier and Support Vector Machine (SVM), for dialogue act tagging in the MmTravel (Myanmar Travel) corpus, and the results of two approaches are slightly different but the result of SVM approach attained in the term of average F-measure scores of 0.79; showed that these approach has moderately good accuracy for Myanmar dialogue.

Keywords—*Myanmar Dialogue Act Recognition (MDAR), Naïve Bayes, Support Vector Machine (SVM), MmTravel corpus*

I. INTRODUCTION

Natural Language Understanding (NLU) is an important component of dialogue management, and NLU has been extraordinarily improved by deep learning techniques, but current NLP techniques for Myanmar language dialogue action classification is new research area where it has been difficult to infer a dialogue act from a surface utterance because it depends on the context of the utterance and speaker linguistic knowledge. From linguistic perspectives on NLU, Allen [15] describes the following forms of knowledge which are phonetic and phonological knowledge, morphological knowledge, syntactic knowledge, semantic knowledge and pragmatic knowledge. Among them, pragmatic knowledge which is focused on how sentences are used in different situations and how to make interpretations of the sentences. In there, one of the subfields of pragmatics, speech act, that studies how words are used not only to present information but also to carry out actions. Dialogue act is also a type of speech act.

Dialogue act recognition is an essential task for dialogue systems. Automatically Dialogue Act Modeling [13][3] and detecting the structure of dialog is critical to better interpret and help in understanding a conversation. The minimal units of linguistic communication of DAs are directly connected with the speaker's communicative intentions. Austin defines the dialogue act is the function of a sentence in the dialogue which means the function of a question is to request some information, while an answer shall provide this information. The former state of act recognition has been addressed by Searle in 1969, which is based on Austin (1962) work as a fundamental concept of linguistic pragmatics, analyzing, for example, what it means to ask a question or make a statement. To the best of our knowledge, there is no research work on Myanmar dialogue act recognition and modeling that have been published in NLP, but there are several works for other languages, especially for English. Different sets of dialogue acts are defined depending on the target application. In [4], standard of discourse structure annotation, the Dialog Act Markup in Several Layers (DAMSL) tag set [14], which designed by the natural-language processing community (Core & Allen) in 1997. In total, 42 dialogue act classes were defined for English. Switchboard-DAMSL [6] tagset is the modified of DAMSL in the telephone conversation domain. And, another popular tagset is the Meeting Recorder Dialogue Act (MRDA) tagset [MRDA], where contains 11 general DA labels and 39 specific labels, which is based on the taxonomy of SWBD-DAMSL. The Map-Task [12] is English tagset, which contains 19 tags. AMI [11], DIT++ [9] and other ISO standards [10] are examples of dialogue act tagset.

Many researchers have been widely explored over the years for dialogue act recognition as a task, using multiple classical machine learning approaches, Hidden Markov Models (HMM) [17], Maximum Entropy (Maxent), Bayesian Network, and Support Vector Machines (SVMs) [5][8]. Recently, the research of DA recognition approaches applies by Neural Network which will be our future work for Myanmar dialogue. The authors proposed a fully-automated method for the task of speech act classification for Arabic discourse, as in [1]. Arabic sentences have been collected from two Arabic news sources: Al-Hayat newspaper and Aljazeera television station, and defined 10 Arabic speech act. Naïve Bayes and Decision Trees algorithms were used to induce speech act classifiers for Arabic texts and used as features by the word-tag pairs. Decision-tree based classifiers worked slightly better for a 3-word context while Naïve Bayes were better for 4- or 5-word contexts. Results obtained using sequences (bigrams and trigrams) of POS tags had higher accuracy when using naïve Bayes classifier but lower or similar accuracy scores when using decision trees. For model evaluation, they evaluated by 10-fold cross-validation. Reference [2] contributed a combination of a Naïve Bayes classifier and n-grams based natural speech dialogue act detection on two corpora: the Switchboard and the Basurde tasks. 66% of accuracy is achieved on the Switchboard corpus by using a uniform Naïve Bayes classifier. And also, it has been used 3-grams and Laplace smoothing to avoid zero probabilities. For the Basurde corpus, they applied Naïve Bayes classifier with 2-grams and Written Bell smoothing and achieved the best accuracy of 89%.

For multiclass DA classification, [16] applied SVMs on the ICSI meeting corpus that consists of 75 natural occurring meetings and each meeting has long an hour by five participants. Act tags are grouped into 5 broad intuitive classes. In their evaluation, they mention combining multiple binary SVMs via error correction output codes (ECOC) achieved better performance than representing a direct multiclass SVM. Reference [7] used the SVM linear kernel combine with HMM models in the HCRC MapTask corpus which is a collection of 128 2-speaker dialogs with twelve tags. The corpus's each dialog has the direction of a shared map from speaker to listener. The authors approved the SVM can easily integrate sparse high-dimensional text features and dense low-dimensional acoustic features. The classification accuracy of 42.5% and 59.1% respectively for acoustic and text features with a linear SVM followed by Viterbi decoding, and 65.5% for combination.

II. MMTRAVEL CORPUS

We proposed Myanmar dialogue corpus which is based on ASEAN MT[16] Myanmar dataset and online resources for travel domain. In our corpus, we also modified ASEAN MT data to an informal conversation between humans, and it has 60k utterances about question and answer conversation. The corpus has longer utterance, on average, there are 30 words and short utterance consists 5 words. The first step in establishing a dialogue act recognition system is defining the appropriate functions or the DA tag-set. Different types of dialogue systems require labeling different kinds of acts. Therefore, we categorized twenty-nine DA tagsets based on five kinds of Myanmar speech function is listed in Table 1 [18], [19], [20], [21], [22], [23], [24].

A. Informational Function

Informational function concentrates on the message between speaker and listener. It is used to give new information and it depends on truth and value. Inform (inf) act is included in this function. It conveys information to make someone aware of something and listener can decide that information is right or wrong. Example: "အကောင်းဆုံး ကော်ဖီမှုန့် သုံးထားတာပါ (the best coffee powder are used)".

B. Expressive Function

The expressive function can be used to express attitudes and feelings. Accept (ac) describes used to agree to take something and to say 'yes' to an offer or invitation. An act of saying sorry is the apology (apol) act. Congratulate (cong) describes when we praise (someone) for an achievement, for example: "ဝိုး တကယ် လား ၊ ဂုဏ်ယူ ပါတယ် (Wow really, congratulation)". Complain act (cp) expresses dissatisfaction or annoyance about something. Deny (dny) act used to not allow someone to have or do something and state that one refuses to admit the truth or existence of, example sentence is "ခင်ဗျား ပြောတာ မဖြစ်နိုင်ဘူး (It's impossible)". Opinion (op) act is a thought or belief about something or someone, like this sentence "ပန်းသီး ဆိုရင် ကောင်းမယ် (Apple is much better)". Our corpus is based on traveling dialogue therefore the shopping and eating is a kind of categories where

review (rev) act to evaluate a service or food. Wish (wn) describes use to hope or express hope for another person's success or happiness or pleasure on an occasion, for example "လမ်းခရီး တစ်လျှောက်လုံး အဆင်ပြေပါစေ (Have a safe journey)".

C. Directive Function

Directive function, which aims to influence the behavior or attitudes of others. There are many kinds of act tag in here:

- Command (cmd) describes acts used to control over someone or something and tell them what to do, for example "မလုပ်နဲ့ (Don't do it)".
- Direction (dir) used to instruct someone about how to find a particular place and pointing (someone) towards.
- Invite (inv) used to invite or request someone to come or go to some places, especially formally or politely.

TABLE I. PROPOSED MYANMAR DIALOGUE ACT TAGSETS

Speech Function	Dialogue Act	Abbrev
Informational	Inform	inf
Expressive	Accept/Agree	ac
	Apology	apol
	Congratulate	cong
	Complain	cp
	Deny	dny
	Opinion	op
	Review	rev
	Wish	w
Directive	Command	cmd
	Directions	dir
	Invite	inv
	Instruction	instr
	Urge	u
	Prohibit	proh
	Request	req
	Suggestion	sug
	Thank	thx
	Warning	wn
	Confirm Question	cfm_q
	Choice Question	ch_q
	Complain Question	cp_q
	Inquiry Question	inq_q
	Request Question	req_q
	Other Question	otr_q
Phatic	Greeting	gt
	Goodbye	gb
	Self_Intro	s_i
Aesthetic	Aesthetic	as

- Instruction (instr) used to advice and information about how to do or use something.
- Urge (u) used to strongly advise or try to persuade someone to do a thing.
- Prohibit (proh) used to officially refuse to allow something.
- Request (req) which is the act of politely or officially asking for something.
- Suggestion (sug) includes an idea, plan, or action that is suggested or the act of suggestion it.
- Thank (thx) describes acts used to express to someone that are grateful for something that they have done.
- Warning (wn) is something that makes you understand there is a possible danger or problem, especially one in the future.
- Question has (6) kind of act tag: Confirm_Q (cfm_q), Choice_Q (ch_q), Complain_Q((cp_q), Inquiry_Q (inq_q), Request_Q (req_q), and Other_Q (otr_q), where we tagged them in sentence or phrase used to find out information.

D.Phatic and Aesthetic Function

Phatic function uses to maintain social relationships, and to begin or continue the conversation, which helps us to interact with people. Greeting (gt), goodbye (gb), and self_intro (s_i) are kind of phatic function. Example: "မင်္ဂလာ နေ့လည်ခင်းပါ ၊ တွေ့ရတာ ဝမ်းသာပါတယ် (Good afternoon, glad to meet you)". The aesthetic function includes the rhythm, balance and contrast of sentence, clauses and words also play.

TABLE II. PERCENTAGE OF UTTERANCES ASSIGNED TO EACH DIALOG ACT CONSIDERED IN THE CORPUS

No.	act_tag	%	Example
1	inf	35.70	ဘာ လို့ လဲ ဆိုရင် ဒီမှာ လတ်ဆတ်တဲ့ အသီးတွေ ကျွန်တော် တွေ့ခဲ့ ပါတယ်
2	inq_q	31.78	ခင်ဗျား လမ်းညွှန်မြေပုံ ပါ လား
3	req	6.90	နောက် တစ်ပတ် သောကြာနေ့ မတိုင်ခင် မန္တလေး မြို့ ကို သွား မယ့် လေယာဉ် မှာ လက်မှတ် တစ်စောင် ကြိုတင် မှာ ချင်လို့ပါ
4	op	5.22	အကောင်းဆုံး လက်ဝတ် ရတနာ ကိုး
5	ac	4.07	အင်း ယူ သွား ပေး မယ်
6	req_q	3.30	ရှင် ကျေးဇူးပြုပြီး ဒါ ကို ပင်မင်းဆိုင် ယူ သွား ပေး မလား

No.	act_tag	%	Example
7	instr	1.40	အဲဒါတွေ ကို ကား ပေါ် တင်လိုက် ပါ
8	thx	1.36	စောင့်ပေး လို့ ကျေးဇူးတင် ပါတယ်
9	apol	1.03	ကျွန်တော် နောက်ကျသွား လို့ တောင်းပန် ပါတယ်
10	dny	1.00	မင်း ကို အဝတ်အစား ဝယ် ဖို့ ငါ တော့ ပိုက်ဆံ တစ်ပြား မှ မ ထုတ်ပေး နိုင် ဘူး
11	sug	1.00	လေဆိပ် ကို မနက် ၈ နာရီ မတိုင်ခင် ရောက်ရင် ကောင်း ပါတယ်
12	gt	1.00	မင်္ဂလာ ပါ ၊ နေကောင်း လား
13	rev	0.84	ကျွန်တော် သိ သလောက်တော့ ဒါဟာ ရှိသမျှ ထဲမှာ တကယ့် အကောင်းဆုံး ပဲ
14	u	0.84	အကောင်းဆုံး ကို မျှော်လင့် ကြ ပါ စို့
15	dir	0.71	ဟိုဘက် လမ်း မှာ စျေးဝယ်စင်တာ ရှိတယ်
16	cp	0.64	မင်း ကြောင့် ခုတော့ ငါ့ မှာ နောက်ထပ် အထုပ် တစ်ထုပ် ထပ်ပို ရဦး မှာ ပေါ့
17	cfm_q	0.57	ခင်ဗျား ကျွန်တော့် ကို တစ်ခုခု ယူလာ စေချင် တာ လား
18	ch_q	0.55	အအေး လား အပူ လား
19	proh	0.45	ဒီနေရာ က ဆေးလိပ် ကင်းစင် နေရာ မို့ ဆေးလိပ် မသောက်ပါနဲ့
20	w	0.43	မင်း ရဲ့ လေကြောင်းခရီး မှာ ပျော်ရွှင် ပါစေ
21	cmd	0.37	တိတ်တိတ် နေ
22	s_i	0.19	ကျွန်မ နာမည် သူဇာ ပါ
23	as	0.18	အလှငံလို အနှေးဖြစ် ဆိုတာလိုပဲ
24	gb	0.15	ကျွန်တော့် ကို ခွင့်ပြု ပါဦး မနက်ဖြန် တွေ့ ပါမယ်
25	inv	0.12	ခင်ဗျား ကို ကျွန်တော်တို့ မိသားစု က အမြဲ ကြိုဆို ပါတယ်
26	wn	< 0.1	နောက် ကို ကျွန်တော် ပြော တဲ့ အတိုင်း လုပ်ပါ ၊ ဒါ နောက်ဆုံး အကြိမ် ဖြစ် ပါစေ
27	cp_q	< 0.1	ဘာ လို့ အခုထိ ရေ မလာသေးတာ လဲ ၊ အဆင်ကို မပြေဘူး
28	cong	< 0.1	ဟုတ် လား ၊ ဂုဏ်ပြု ပါတယ် နော်
29	otr_q	< 0.1	မင်း ဘာ ကို မ ဝယ် ခဲ့ သလဲ ဆိုတာ

Aesthetic (as) tag example sentence is "လူပြောမသန် လူသန်မပြော".

The most top five frequent DA types include INFORM, INQUIRY QUESTION, REQUEST, OPINION, and AGREEMENT/ACCEPT. We list of all dialogue acts in our corpus ordered by the highest frequencies together with the example sentence can be seen in Table 2, where Myanmar sentences translation are given in Appendix A. Generally, our daily conversation tends to give information about something and implies the imparting of knowledge especially of facts or occurrences to the listener, that has been made the INFORM (inf) act tag to increase the percentage.

III. METHODOLOGY

Dialogue act classification is a special case of text classification, where the samples are the utterances, and the attributes are the words of that.

A. Naïve Bayes Classifier

The supervised learning method, Bayesian Classification, is also a statistical method to classify document or text. Its algorithm based on applying Bayes theorem with naïve assumption which means every feature is independent of the others, in order to predict the category of a given sample. The probabilistic model calculates the probability of each category using Bayes theorem, and categorize with the highest probability will be output. To find a function f*(.) which maps an utterance u_j into an act label a_j. The range is defined by the number of act labels A, where includes 29 dialogue acts of the MmTravel task. In the training, the learning function is done from a set of samples with the form:

$$\{(u_j, a_j)\}_{J=1}^{J},\ u_j \in U,\ a_j \in A \qquad (1)$$

where u_j is the j-th sample, J is the number of samples, and a_j correspond act label. The decision of act tag is assigned to each utterance, which is made by the Bayes decision rule for minimizing error. The classifier assigns the act with maximum probability to the utterance u:

$$a_{MAP} = \underset{a \in A}{\text{argmax}}\ P(a|u) \qquad (2)$$

where MAP is "maximum posteriori" which means most likely class are assigned the result. By Bayes rule,

$$= \underset{a \in A}{\text{argmax}}\ \frac{P(u|a)P(a)}{P(u)} \qquad (3)$$

For dropping the denominator,

$$= \underset{a \in A}{argmax}\ P(u|a)P(a) \qquad (4)$$

$$a_{MAP} = \underset{a \in A}{\text{argmax}}\ P(x_1, x_2, ..., x_n|a)P(a) \qquad (5)$$

where utterance u represented as features $x_1, x_2, \ldots, x_n$.

B. Support Vector Machine

SVM is one of the robust classification models with good generalization ability on unseen data, and it is also a binary classifier where it works on the concept of identifying the separating hyper plane, as decision boundary, with maximum margin between two classes. The objective of SVM is to define the decision boundary with the maximum margin, where it can be defined the distance from the nearest training example to the decision boundary. SVM classifier can be used to classify not only two given classes but also multiple classes.

The basic classifier is linear classifier which separate a linearly separable data. In non-linear decision boundary SVM draw boundary by transforming the input original space to a high dimensional space. The function transforming data to a non-linear space is called Kernel functions. If the classifier can determine the optimal separating hyper plane, it guarantees to give better result for its model. The decision boundary is determined by the discriminant:

$$f(x) = \sum_i a_i \lambda_i K(u,u_i) + b \qquad (6)$$

where u_i and $a_i \in \{-1,1\}$ are the utterance-tag pairs for training set $\{(u_i, a_i)\}_{i=1}^{N}$. The kernel function, $K(u, a) \doteq \phi(u) . \phi(a)$, which computes inter products, and $\phi(u)$ is a transformation from the input space to a higher dimensional space. $\phi(u) = u$ used for the linearly separable case. In non-linear separable cases, an SVM apply the mapping $\phi(.)$ to increase dimensionality and after that applying a linear classifier in the higher dimension. Kernel functions $K(. , .)$ of SVM classifier are listed below:

For Linear Kernel,

$$K(u,u_i) = u_i^T u \qquad (7)$$

For Polynomial Kernel,

$$K(u,u_i) = (u_i^T u + \tau)^d \qquad (8)$$

For Radial Basis Function (RBF) Kernel,

$$K(u,u_i) = \exp(- \| u - u_i \|_2^2 / \sigma^2) \qquad (9)$$

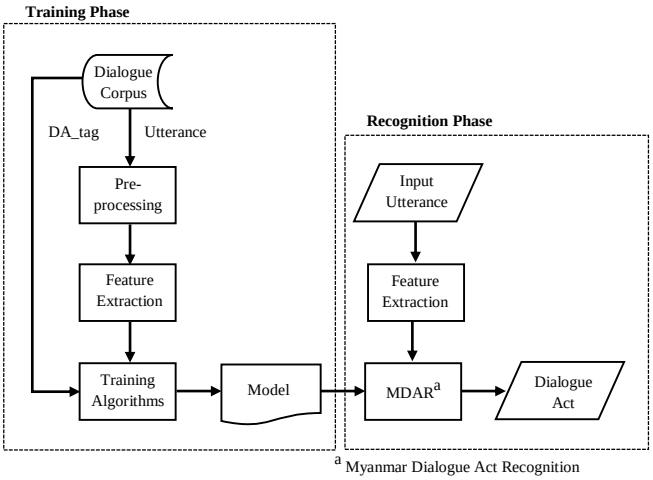

Figure 1. Flow chart of the proposed Myanmar Dialogue Act Recognition

IV. MYANMAR DIALOGUE ACT RECOGNITION

Dialogue act recognition task is the same as classification which acts a dialogue label to sentence, where multiple classification methods have been used to do this undertaking. Most of them were supervised approaches, which means that acquired a large amount of annotated data to obtain models. For this reason, we firstly prepared MmTravel corpus which coverage 12 categories, where includes transportation, tourism, sightseeing, shopping, healthcare, food, emergency, communication, airport, accommodation and other online resources for travel domain. The illustration of MDR proposed flow chart is shown in Fig. 1.

A. Pre-processing

In pre-processing phase, we applied Myanmar word segmentation tool[1] that published by UCSY[2] NLP Lab. We split the dataset by using cross-validation with 80% for the training set and 20% in the test set to perform the experiments.

B. Feature Extraction

Feature extraction used to extract features from datasets to a format supported by machine learning approaches. DA classification is very similar to sequence classification problem. Most of the other languages dialogue act detection system uses the feature-based statistical classification combine with several feature such as linguistic and prosody, by using appropriate machine learning approaches. But in our work, we use tfidfvectorizer (term frequency-inverse document frequency vectorizer) which turning a

[1] http://www.nlpresearch-ucsy.edu.mm/NLP_UCSY/wsandpos.html
[2] University of Computer Studies, Yangon – www.ucsy.edu.mm

collection of raw data into numerical feature vectors and show the resulting scores assigned to each word. Term frequency summarizes how often a given word appears within a document, and inverse document frequency downscales words which appear a lot across documents. The statistical measure, tf-idf weight: $w_{i,j} = tf_{i,j} \times \log(\frac{N}{df_i})$, used to evaluate which word is important in an utterance in a corpus.

C. Training

For the training algorithm, we analyzed MultinomialNB and BernoulliNB for Naïve Bayes classifier, and different kernel-based SVM machine learning on our annotated data. Some of the top features extracted by MultinomialNB and BernoulliNB classifier on our datasets are listed in Table 3 and 4 respectively. Among them, some of the common words in

TABLE III. TOP FEATURES BY MULTILNOMIAL NAÏVE BAYES

Dialogue Act		Features Words
Request Question	req_q	ဝမ်းနည်း ၊ နိုင်မလား ၊ လုပ်ပေး ၊ ပေးပါ
Directions	dir	ဒီကား ၊ ဒီလမ်း ၊ ဒီမှာပဲ ၊ ဒီည ၊ ဒီညနေ ၊ ဒီထက် ၊ ဒီနား ၊ ဒီနေရာ ၊ ဒီဘက်
Command	cmd	ထည့် ၊ ဖို့ ၊ ရဘူး ၊ နော် ၊ တော့ ၊ ကို ၊ မ ၊ နဲ့
Greeting	gt	ဟဲလို ၊ မင်္ဂလာ ၊ **ကျွန်တော်** ၊ ဟိုင်း ၊ ဝမ်းသာ ၊ ရဲ့ ၊ နေကောင်း ၊ **ခင်ဗျား**

TABLE IV. TOP FEATURES BY BERNOULLI NAÏVE BAYES

Dialogue Act		Features Words
Instruction	instr	ရမယ် ၊ သောက် ၊ ပါ ၊ **မင်း** ၊ က ၊ ဖို့ ၊ ဒီ ၊ နဲ့ ၊ ကို
Apology	apol	အားနာ ၊ စိတ်မကောင်းပါဘူး ၊ တောင်းပန်
Deny	dny	မဟုတ်ပါဘူး ၊ ဘူး ၊ မဟုတ်ဘူး ၊ မ ၊ ဟင့်အင်း ၊ နဲ့
Wish	w	မျှော်လင့် ၊ ဟော်ရှင် ၊ မှာ ၊ ခရီး ၊ **ကျွန်တော်** ၊ ပါစေ

Myanmar language as ကျွန်တော် (“I”), မင်း (“You”), ခင်ဗျား (“You”), are appear frequently in the corpus. So, we will be estimated classification by deleting stopwords from the utterances in our future work.

Support Vector Machine (SVM), as a powerful supervised learning method, is suitable to deal with classification problems and has high accuracy rate for linearly inseparable data, which can be mapped into a high dimensional space through kernel functions. In our experiment, we study the impact of kernel functions which are linear kernel, RBF kernel, and Polynomial kernel. Our experiment among these three kernels, the learning ability of RBF kernel is stronger than the other kernels, we also express the training and evaluation report for RBF kernel as follows:

Accuracy on Training Set : 0.8568779
Accuracy on Testing_Set : 0.7913935
နောက်ထပ် ရော ဘာ ဖြစ်သေး လဲ
Predicted Target: inq_q
Actual Target: inq_q
မင်း အချိန် ရှိလာ မှာ ပါ
Predicted Target: inf
Actual Target: op
ဒါလေး ယူသွား ပေးပါ လား
Predicted Target: req_q
Actual Target: req_q
ခင်ဗျား စီးချင် တဲ့ လေကြောင်းလိုင်း ကို ခေါ် ပြီး
ခင်ဗျား ဘာတွေ သယ်နိုင် လဲ ဆိုတာ မေးနိုင် ပါတယ်
Predicted Target: inq_q
Actual Target: u
အခုလို တွေ့ ရတာ ဝမ်းသာ ပါတယ်
Predicted Target: gt
Actual Target: gt
ဒီလမ်း အတိုင်း တည့်တည့် လျှောက် ပါ ပြီးရင် ညာကွေ့
Predicted Target: dir
Actual Target: dir

[3]Example sentences translation are given in Appendix B

Table 5. shows the classification report for (29) dialogue act tags by using Precision (P), Recall (R), and the F1 score (F1). But the result for act tag: as, cong, gb, inv, and otr_q, are less than 0.1% because of the small number of these tags contained in our corpus, and we will be considered the data balancing for all tagsets in our future work.

D. Experiment

We presented the classification result of the Naïve Bayes and each kernel type of SVM classifier for the DA classification task is shown in Table 6. Overall, from the range of classification undertaken, SVM RBF kernel was obtained the highest average classification accuracy at 79 % compares favorably to the other classifiers and it performs slightly better than linear and polynomial kernels, suggesting that the RBF kernel approach may be able to capture some discriminative features that directly distinguish different classes.

TABLE V. CLASSIFICATION RESULT OF RBF KERNEL

Dialogue Act		Precision	Recall	F1
Accept/Agree	ac	0.73	0.65	0.68
Apology	apol	0.90	0.77	0.83
Aesthetic	as	< 0.1	< 0.1	< 0.1
Confirm Question	cfm_q	0.53	0.15	0.24
Choice Question	ch_q	0.72	0.34	0.46
Command	cmd	0.83	0.14	0.24
Congratulate	cong	< 0.1	< 0.1	< 0.1
Complain	cp	1.00	0.08	0.14
Complain Question	cp_q	0.50	0.20	0.29
Directions	dir	0.78	0.27	0.40
Deny	dny	0.84	0.47	0.60
Goodbye	gb	< 0.1	< 0.1	< 0.1
Greeting	gt	0.87	0.58	0.69
Inform	inf	0.72	0.93	0.81
Inquiry Question	inq_q	0.88	0.93	0.91
Instruciton	instr	0.70	0.09	0.15
Invite	inv	< 0.1	< 0.1	< 0.1
Inquiry Question	iq_q	0.83	0.35	0.50
Opinion	op	0.61	0.36	0.46
Other Question	otr_q	< 0.1	< 0.1	< 0.1
Prohibit	proh	0.53	0.19	0.28
Request	req	0.86	0.78	0.82
Request Question	req_q	0.90	0.72	0.80
Review	rev	0.45	0.06	0.10
Self_Intro	s_i	0.83	0.42	0.56
Suggestion	sug	0.84	0.13	0.23
Thank	thx	0.87	0.84	0.86
Urge	u	0.87	0.57	0.69
Warning	wn	0.93	0.53	0.68

TABLE VI. DIALOGUE ACT CLASSIFICATION SCORE FOR NAÏVE BAYES AND DIFFERENT KERNEL OF SVM CLASSIFIER

	Precision	Recall	F-score
SVC Linear kernel	0.768	0.781	0.758
SVC RBF kernel	**0.796**	**0.815**	**0.787**
SVC Poly kernel	0.771	0.753	0.727
MultinomialNB	0.728	0.729	0.696
BernoulliNB	0.73	0.739	0.733

V. CONCLUSION AND FUTURE WORK

In this work, we have explored the use of machine learning based MDAR, which can recognize the sequence of utterances in a conversation with multiclass DA classification on MmTravel corpus. Experimental results highlight that the SVM RBF kernel is better than the other approaches. Our perspective for the near future is to improve our MmTravel corpus from 60k to 100k, and also be aware of the dialogue act tag unbalancing data which will be overcome the zero F1 score for some act tags. Nowadays, every research has a scope for improvement with deep neural networks. As a future extension to this work, we are planning to analyze the performance of our MDAR with sequence learning neural networks.

APPENDIX A

1. (Because I have found fresh fruits here)
2. (Do you have a map?)
3. (I want to book an air ticket to Mandalay until next Friday)
4. (The best jewelry)
5. (Ok, I will take it)
6. (Could you please bring it to the laundry?)
7. (Put them in the car)
8. (Thanks for waiting)
9. (I apologize for being late)
10. (I can't give you any money to buy clothes)
11. (It is better to arrive at the airport before 8 am)
12. (Good morning, how are you?)
13. (As far as I know, this is the best of all)
14. (Hope for the best)
15. (There is a shopping center next street)
16. (Now I have to sent another packet because of you)
17. (Do you want me to bring something?)
18. (Hot or cold)
19. (Don't smoke, this is non-smoking area)
20. (Enjoy your flight!)
21. (Silent!)
22. (I'm Thuzar)
24. (Let me go, see you tomorrow)
25. (We always welcome you)
26. (This is the last time to you, do what I say next time)
27. (Why font is not working, it is inconvenient?)
28. (Really, congratulation)
29. (Why you didn't buy?)

APPENDIX B

နောက်ထပ် ရော ဘာ ဖြစ်သေး လဲ (What else happened)
မင်း အချိန် ရှိလာ မှာ ပါ (You will have time)
ဒါလေး ယူသွား ပေးပါ လား (Please take this)
ခင်ဗျား စီးချင် တဲ့ လေကြောင်းလိုင်း ကို ခေါ် ပြီး ခင်ဗျား ဘာတွေ သယ်နိုင် လဲ ဆိုတာ မေးနိုင် ပါတယ် (Contact the airline you want to take and ask what you can bring)
အခုလို တွေ့ ရတာ ဝမ်းသာ ပါတယ် (Nice to meet you too)
ဒီလမ်း အတိုင်း တည့်တည့် လျှောက် ပါ ပြီးရင် ညာကွေ့ (Go straight and turn right)

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[18] ဒေါက်တာခင်အေး၊ အတ္ထပေဒနိဒါန်း

[19] ဒေါက်တာအောင်မြင့်ဦး၊ လူမှုဘာသာဗေဒ မိတ်ဆက်

[20] ဒေါက်တာအောင်မြင့်ဦး၊ လူမှုဘာသာဗေဒ သဘောတရား

[21] မမြင့်မြင့်စန်း၊ မြန်မာဘာသာစကားတွင် ဘာသာစကားအသုံးနှုန်း အနက်အဓိပ္ပာယ်ဆက်သွယ်မှု၊ ပါရဂူဘွဲ့အတွက် တင်သွင်းသော ကျမ်း၊ မြန်မာစာဌာန၊ ရန်ကုန်တက္ကသိုလ်

[22] မဥမွာစိုး၊ မြန်မာဘာသာစကား၌ လမ်းညွှန်မှု အသုံးများ၊ ပါရဂူဘွဲ့အတွက် တင်သွင်းသောကျမ်း၊ မြန်မာစာဌာန၊ ရန်ကုန်တက္ကသိုလ်

[23] မောင်ခင်မင်(ဓနုဖြူ)၊ လက်တွေ့ အတ္ထပေဒနိဒါန်း

[24] မောင်ခင်မင်(ဓနုဖြူ)၊ အတ္ထပေဒနိဒါန်း

Myanmar News Retrieval in Vector Space Model using Cosine Similarity Measure

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Abstract

Information Retrieval (IR) is an effective means of retrieving the best relevant document to user query. Nowadays, the problem of documents similarity deals with IR is retrieving required information from a large amount of data. In this paper we studied a Vector Space Model (VSM) that is used in IR and represents a document as a vector in an n-dimensional space, where each dimensional represents a term and measured through cosine angle between two vectors. The objective of this paper is to retrieve the relevant file from Myanmar news data sets using cosine similarity measure in VSM to user's query. Evaluations are done in terms of similarity score by Precision, Recall and F-score.

Keywords— *Information Retrieval, TF-IDF weighting scheme, cosine similarity measure, Vector Space Model*

I. INTRODUCTION

The basic form of Information Retrieval (IR) is based on an input query for retrieving of documents. Nowadays, many users are searched documents on the web. The complete example for this application is web search. Web search engines are the most visible in IR applications. For this purpose, developed many algorithms accept a user query and search it in the documents collection and rank the results of similarity score relevant to the user query. Against the individual query terms, these algorithms based on maintain the information deals with term frequencies and positions which matched to indexed documents. Assigning to each document focused on its value is similarity score. High for a high frequency in the document is the query term's score. In analyzing this similarity and computing the score, different algorithms take different approaches. To do this work, the Vector Space Model (VSM) is one of the best approaches. It overcomes the Boolean Model, which uses Boolean desired queries based on Boolean logic and searches it in the documents, retrieval result is based on either the desired terms are in the documents or not. This gives too many or too few documents [9]. The role of textual information retrieval is term weighting. Term Frequency and Inverse Document Frequency (TF/IDF) weighting scheme is one of the most popular schemes of term weighting, followed by Okapi BM25. We discuss three modified schemes in the following text. A new term weighting scheme, Term Frequency with Average Term Occurrences (TF-ATO), [3] which is an advance than the TF/IDF weighting scheme. These schemes use a discriminative approach depend on the document based vector to discard fewer significant weights from the documents and calculate the average term occurrences of terms in documents. TF-ATO effect of discriminative approach and the stop words removal process on the IR system capability and achievement than using the well-known TF-IDF that is an importance part in the Vector Space Model (VSM). Another modified TF-IDF scheme [4] which exploits two features of within document term frequency normalization to decide the critical of a term. One component of the TF tends to adopt short documents, while the other tends to adopt long documents and combine these two TF components using the query length information that keeps a balanced trade-off in retrieving short and long documents, when the ranking function allows queries of different lengths. The IDF of a term is similar distance between the empty string and the term which is approximated shown that M. Shirakawa [5]. They have proposed a global term weighting technique, N-gram IDF, by collaborating IDF and described the clarity and durability of N-gram IDF on key term abstraction and Web search query tokenization functions. It able to achieve competitive performance with advanced methods planned for any task adding efforts and assets.

II. MYANMAR INFORAMTION RETRIEVAL

One of the Natural Language Processing (NLP) advanced techniques, Information retrieval (IR) is defined to be the science of enhancing the effectiveness of term-based document retrieval. An IR system is an information system that collects, indexes and keeps the data for extracting of relevant information responding to a user's query (user information need). An IR process begins when a user enters a query into the system. Queries are formal statements of information needs, for example search strings in web search engines. IR is searching material (usually documents) of an unstructured nature (usually text) that satisfies an information need from within large. In this paper, we used Myanmar information retrieval for retrieving file containing Myanmar news documents from Myanmar news corpus that is relevant to user query.

III. DATA COLLECTION

In this part, we collects Myanmar news corpus about 7000 documents written with Myanmar Unicode font from Mizzima Burmese news website. Types of news are shown in Table 1.

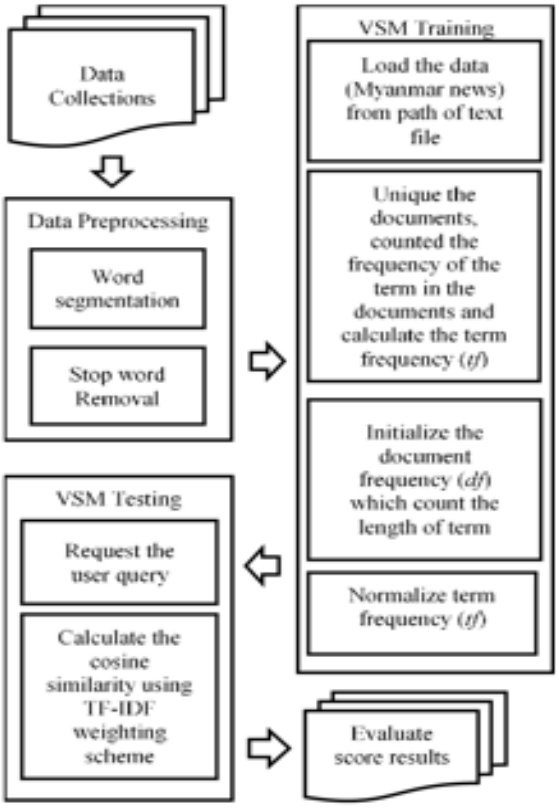

Figure 1. The process of Myanmar news retrieval

TABLE I. STATISTIC OF MYANMAR NEWS CORPUS

Type of news	Number of documents	Number of documents
Health	2115	18195
Sport	1349	2865
Entertainment	676	6277
Polictical	1423	15538
Economic	1437	11836
Total	7000	54711

IV. MYANMAR NEWS RETRIEVAL

Myanmar online news was collected for News Corpus from Myanmar news website and was done the word segmentation and stop words removal as in data preprocessing steps. Vector Space Model (VSM) using cosine similarity measure is applied in this work to train and test the corpus for the final output is the similarity score that are relevant to user query, and the process is shown in Fig. 1. We used about 700 corpus in training phrase and about 4000 corpus in testing phase. Preprocessing steps is described in section V with detail description of training phase with four modules: loading the Myanmar news data collection, unique the documents in data collection and counting the term *t* occurrences in the documents *d* and calculation the term frequency (*tf*), Initializing the document frequency (*df*) that is the document *d* occurrences in the term *t* and the last is normalized term frequency (*tf).* Testing phase has contained two parts: one is requested the user query and tokenized these query and then calculated the cosine similarity measure using TF-IDF weighting scheme and the final results is relevant to user query.

V. DATA PREPROCESSING

In this step, we collected Myanmar news data from Mizzima Burmese news website. All of this news is used Zawgyi fonts. Therefore, we transformed to Unicode fonts, tokenized this news and removed the stop words as described below [13].

A. Word segmentation

Word segmentation is the fundamental task in natural language processing. This process split into word and sentences that are searching word tokens and borders of sentences. English word boundaries can be

easily defined but it is not in Myanmar. In Myanmar word boundaries, words are frequently written without spacing in sentences. Therefore, for pass sentences and word tokens, word segmentation is very useful for Information Retrieval (IR). For this purpose, we used Myanmar word segmenter of UCSY [10]. In preprocessing steps, word segmentation is very important for evaluation of IR.

B. Stop word removal

The objective of stop word removal is filter out words that appear in most of the documents. The examples of stop words removal are also shown in Table 2 [12]. This step is very important step in preprocessing techniques used in Natural Language Processing application.

TABLE II. EXAMPLE OF PREPROCESSING SENTENCES

Original sentences	Preprocessed sentences
ဧရာဝတီတိုင်း တွင် တွေ့ ရှိခဲ့ သော ဒုက္ခသည် များ ကို ရခိုင်ပြည်နယ် တွင် ယာယီ ထား‌ရှိ ရေး	ဧရာဝတီတိုင်း
	တွေ့ ရှိ
	ဒုက္ခသည်
	ရခိုင်ပြည်နယ်
	ယာယီ ထား‌ရှိ ရေး
အမျိုးသား လူ့ အခွင့်အရေး ကော်မရှင် ထံ တစ်ပတ် အတွင်း တိုင်ကြား သွား‌မည်	အမျိုးသား လူ့ အခွင့်အရေး ကော်မရှင်
	တစ်ပတ်
	တိုင်ကြား
	သွား
ရွေးကောက်ပွဲ ကော်မရှင် ၏ လုပ်ဆောင်ချက် များ သည် ဥပဒေ နှင့် မ ညီ သည့် အမှား များ အပြင် မ မှား သင့် သည် များ ကိုပါ မှားယွင်း နေသည် ဟု ဒေါ်အောင်ဆန်းစုကြည် ဦးဆောင်သည့် အမျိုးသား ဒီမိုကရေစီ အဖွဲ့ချုပ် (NLD) က ဖွန်လ ၃ ရက် တွင် အိတ် ဖွင့် ပေးစာ တစ်စောင် ပေးပို လိုက် သည် ။	ရွေးကောက်ပွဲ ကော်မရှင်
	လုပ်ဆောင်ချက်
	ဥပဒေ
	မ ညီ
	အမှား
	မ မှား သင့်
	မှားယွင်း
	အောင်ဆန်းစုကြည်
	ဦးဆောင်
	အမျိုးသား ဒီမိုကရေစီ အဖွဲ့ချုပ်
	NLD
	ဖွန်လ ၃ ရက်
	အိတ် ဖွင့် ပေးစာ
	တစ်စောင်
	ပေးပို
ရှမ်းပြည်နယ် တောင်ပိုင်း ပြည်သူ ဖိရမ် တွင် ဖက် ဒရယ် မူ နှင့်ပတ်သက်ပြီး ဒေသခံ ပြည်သူများ နှင့် အရပ်ဘက် အဖွဲ့အစည်းများ သေချာစွာ သိရှိ နား‌လည် စေရန် ဆွေးနွေးပွဲ တစ်ခု ထည့်သွင်း ဆွေးနွေး သွား‌မည်	ရှမ်းပြည်နယ် တောင်ပိုင်း ပြည်သူ ဖိရမ်
	ဖက် ဒရယ် မူ
	ပတ်သက်
	ဒေသခံ ပြည်သူ
	အရပ်ဘက် အဖွဲ့အစည်း
	သေချာ
	သိရှိ နား‌လည်
	ဆွေးနွေးပွဲ တစ်ခု ထည့်သွင်း ဆွေးနွေးသွား
	ယခုနှစ် ကုန် ပိုင်း
ယခုနှစ် ကုန် ပိုင်း အတွင်း ကျင်းပ ရန် လျာ ထား‌သည့် အထွေထွေရွေးကောက်ပွဲ တွင် အစိုးရအဖွဲ့ က အဖွဲ့အစည်း အသီးသီး နှင့် ပူးပေါင်း မည်	ကျင်းပ
	လျာ ထား
	အထွေထွေရွေးကောက်ပွဲ
	အစိုးရအဖွဲ့
	အဖွဲ့အစည်း အသီးသီး
	ပူးပေါင်း

VI. VECTOR SPACE MODEL

The Vector Space Model (VSM) represents documents and queries as vectors in multidimensional space, whose dimensions are terms used to build an index to represent the documents. In information retrieval used the VSM, indexing and relevancy rankings and can be successfully used in evaluation of web search engines. The VSM procedure can be divided into three steps. The first is the document indexing where content carrying terms are extracted from the document. The second is the weighting of the indexed terms to improve retrieval of documents relevant to the user. The last step ranks the document with respect to the query according to a similarity measure. A common similarity measure known as cosine determines angle between the query vector and the document vector as described in next section. The angle between two vectors is considered as a measure of difference between the vectors, cosine angle is used to calculate the numeric similarity, determines angle between the query vector and the document vector. The VSM is an algebra model for characterizing text documents which outperform the Boolean Model limitations. The major advantage of VSM is used the weights applied to the term which not binary. VSM grants for more effectively eliminating results and evaluating over similarity range values than the Boolean Model. This model means vectors of weights as documents and queries. Each weight is a measure of the important of an index term in a document or a query, commonly. The index term weights are applied on the basic of the density of the index terms in the document, the query or the collection. As a further process, the document can be decreases different word forms into a useful stem which provides improve the matching the similarity between each documents. As to give better results, the query terms perhaps weighted assign to their related importance. This paper tends to relevance the file in documents from the user query.

A. Cosine Similarity Measure

In Vector Space Model (VSM), the query and documents are represented by a two-dimensional vector. The vector cosine measure is a useful method

to measure the similarity between the two vectors. It is measured by the cosine of the angle between the two vectors and determines whether two vectors are pointing in the same direction (Fig. 2). For considerate this similarity between the two vectors, Euclidean distance is a bad measure and it does not give for realistic results. The two vectors are at 90 degrees to each other is 0 for the cosine value and it has no match. The smaller the angle and the greater the relevant between vectors is near the cosine value to 1. If the documents are similar, evaluation of 0° angle is 1 of similarity score and if the documents are also entirely dissimilar, evaluation of angle 90° is 0 of similarity score.

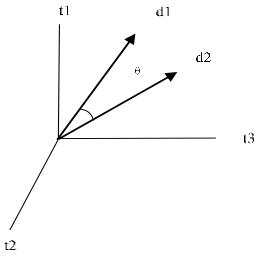

Figure 2. Illustration of angle similarity between two documents

For comparing their weights, implementation of length normalized vectors is the cosine weighting measure. For Cosine Similarity gives the formula, as in (1).

$$Similarity = cos(d', q') = \frac{d'.q'}{||d'||.||q'||} = \frac{\sum_{i=1}^{|V|} d_i q_i}{\sqrt{\sum_{i=1}^{|V|} d_i^2} \sqrt{\sum_{i=1}^{|V|} q_i^2}} \quad (1)$$

B. Term Weighting Scheme

The term frequency, $tf_{t,d}$ is times of the term t appears in document d. The term t computes the document-query match scores. The log-frequency weighting with term in the documents is defined as,

$$W_{t,d} = 1 + log_{10}(tf_{t,d}) \quad (2)$$

Rare in the collection for a term in a query that is a document. To apply for this, document frequency, df_t is the document d occurrences appear anywhere occurs the term t. idf_t means an inverse document frequency is also usefulness of the term t.

$$\text{idf}_t = \log_{10}(\frac{N}{df_t}) \quad (3)$$

The *tf-idf* is the product together with *tf* weight and *idf* weight that is one of the perfect weighting schemes in the information retrieval.

$$W_{t,d} = (1 + log_{10}(tf_{t,d})) * log_{10}(\frac{N}{df_t})) \quad (4)$$

It increases in a term occurrences appearing a document and the term query rarity in the documents collection.

VII. Experiments and Experimental Results

This paper evaluated Myanmar news data sets over 7000 documents and used Vector Space Model (VSM) with cosine similarity measure. As our experiments, we used the VSM python module to retrieve documents automatically. The similarity score results retrieved from Myanmar news documents corpus that is relevant to user query.

As our experimental results, the score of similarity results are relevant to user query. In preprocessing steps, we initially collects Myanmar news corpus about 7000 documents from Mizzima websites and used Myanmar word segmenter of UCSY [10] for Myanmar sentences and word segmentation. In our experimental results, we prepared Myanmar words from collected Myanmar news corpus for user query to request their required information. As in Fig. 3, we randomly used one to nine words user queries about 300 words search, T1-T3 described one to three words about 100 words search, T4-T6 also described four to six words and T7-T9 also described seven to nine words about 200 words search. The Fig. 3 shows the average Precision, Recall and F-score value obtained by calculating Vector Space Model (VSM) for 300 user queries. The VSM rank the documents with their similarity score. Each score calculated by VSM measured with cosine similarity. The documents having zero scores are irrelevant and assigned the lowest possible rank. Therefore, their ranks are required for calculating Precision, Recall and F-score. These values are used in measure the relevancy of documents. The results of Fig. 3 for retrieving the top documents as our threshold value and calculated by Precision, Recall and F-score. As shown in Fig. 3, the maximum and minimum of Myanmar words queries are present in the numbers of words length.

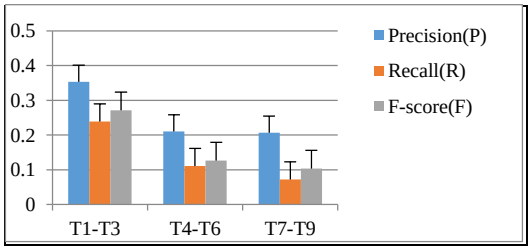

Figure 3. Similarity results used cosine in vector space model for user query

The maximum numbers of T1-T3 means one to three words search queries and the minimum numbers of T7-T9 also means seven to nine words search queries. As this result, we studied that the ranking of documents can vary with words search query length because of their weighting scheme. So, T1-T3 is highest rate in other query length and T7-T9 is also lowest rate in other query length. The final results are obtained the similarity score of T1-T3, T4-T6 and T7-T9 in Precision, Recall and F-score respectively shown in Fig 3.

VIII. CONCLUSION AND FUTURE WORK

In this paper analyzed approach of Vector Space Model (VSM) for check retrieval queries. The similarity value is calculated by using approach of VSM. After analyzing the weighting terms in document collection, was evaluated by similarity value between queries and documents. Documents ranking based on the score of similarity value evaluated by VSM approach. Experiments applied on Myanmar news data sets and the proposed model show that outperforms relevant file in Myanmar news data sets to user query. In this field, future work would be developing new similarity measures, new weighting schemes and new models that can efficiently focused on a huge amount of data sets utilizing semantic information.

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Neural Machine Translation between Myanmar (Burmese) and Dawei (Tavoyan)

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Abstract

This work explores the first evaluation of the quality of neural machine translation between Myanmar (Burmese) and Dawei (Tavoyan). We also developed Myanmar-Dawei parallel corpus (around 9K sentences) based on the Myanmar language of ASEAN MT corpus. We implemented two prominent neural machine translation systems: Recurrent Neural Network (RNN) and Transformer with syllable segmentation. We also investigated various hyper-parameters such as batch size, learning rate and cell types (GRU and LSTM). We proved that LSTM cell type with RNN architecture is the best for Dawei-Myanmar and Myanmar-Dawei neural machine translation. Myanmar to Dawei NMT achieved comparable results with PBSMT and HPBSMT. Moreover, Dawei to Myanmar RNN machine translation performance achieved higher BLEU scores than PBSMT (+1.06 BLEU) and HPBSMT (+1.37 BLEU) even with the limited parallel corpus.

Keywords**:** *neural machine translation, Myanmar-Dawei parallel corpus, Recurrent Neural Network and Transformer*

I. INTRODUCTION

The Myanmar language includes a number of mutually intelligible Myanmar dialects, with a largely uniform standard dialect used by most Myanmar standard speakers. Speakers of the standard Myanmar may find the dialects hard to follow. The alternative phonology, morphology, and regional vocabulary cause some problems in communication. Machine translation (MT) has so far neglected the importance of properly handling the spelling, lexical, and grammar divergences among language varieties. In the Republic of the Union of Myanmar, there are many ethnical groups, and dialectal varieties exist within the standard Myanmar language. To address this problem, we are developing a Myanmar (Burmese) and Dawei (Tavoyan) parallel text. We conducted statistical machine translation (SMT) experiments between Myanmar and Dawei and obtained the highest BLEU scores 45.58 BLEU score for Myanmar to Dawei and 63.22 BLEU score for Dawei to Myanmar [25]. Deep learning revolution brings rapid and dramatic change to the field of machine translation. The main reason for moving from SMT to neural machine translation (NMT) is that it achieved the fluency of translation that was a huge step forward compared with the previous models. In a trend that carries over from SMT, the strongest NMT systems benefit from subtle architecture modifications and hyperparameter tuning.

NMT models have advanced the state of the art by building a single neural network that can learn representations better [4]. Other authors conducted experiments with different NMTs for less-resourced and morphologically rich languages, such as Estonian and Russian [5]. They compared the multi-way model performance to one-way model performance, by using different NMT architectures that allow achieving state-of-the-art translation. For the multiway model trained using the transformer network architecture, the reported improvement over the baseline methods was +3.27 BLEU points. Honnet et al., 2017 [6] proposed solutions for the machine translation of a family of dialects, Swiss German, for which parallel corpora are scarcee. The authors presented three strategies for normalizing Swiss German input to address the regional and spelling diversity. The results show that character-based neural machine translation was the most

promising strategy for text normalization and that in combination with phrase-based statistical machine translation it achieved 36% BLEU score.

In their study, NMT outperformed SMT. In our study, we performed the first comparative NMT analysis of Myanmar dialectal language with two prominent architectures: recurrent neural network (RNN) and transformer. We investigated the translation quality of the corresponding hyper-parameters (batch size, learning rate and cell type) in machine translation between the standard Myanmar and Dawei (one of the dialects) languages. We used syllable byte pair encoding (Syllable-BPE) segmentation for all NMT experiments. In addition, we compared the performance of SMT and NMT experiments with the RNN and transformer.

We found that LSTM cell type with RNN architecture is the best for Dawei Myanmar and Myanmar-Dawei neural machine translation. Myanmar to Dawei NMT achieved comparable results with PBSMT and HPBSMT. Moreover, Dawei to Myanmar RNN machine translation performance achieved higher BLEU scores than PBSMT (+1.06 BLEU) and HPBSMT (+1.37 BLEU) even with the limited parallel corpus.

II. Related Work

Karima Meftouh et al. built PADIC (Parallel Arabic Dialect Corpus) corpus from scratch, then conducted experiments on cross dialect Arabic machine translation [10]. PADIC is composed of dialects from both the Maghreb and the Middle-East. Some interesting results were achieved even with the limited corpora of 6,400 parallel sentences.

Using SMT for dialectal varieties usually suffers from data sparsity, but combining word-level and character-level models can yield good results even with small training data by exploiting the relative proximity between the two varieties [11]. Friedrich Neubarth et al. described a specific problem and its solution, arising with the translation between standard Austrian German and Viennese dialect. They used hybrid approach of rule-based preprocessing and PBSMT for getting better performance.

Pierre-Edouard Honnet et al. [6] proposed solutions for the machine translation of a family of dialects, Swiss German, for which parallel corpora are scarce. They presented three strategies for normalizing Swiss German input in order to address the regional and spelling diversity. The results show that character-based neural MT was the most promising one for text normalization and that in combination with PBSMT achieved 36% BLEU score.

III. Dawei Language

The Tavoyan or Dawei dialect of Burmese is spoken in Dawei (Tavoy), in the coastal Tanintharyi Region of southern Myanmar (Burma). The large and quite distinct Dawei or Tavoyan variety (tvn) is spoken in and around Dawei (formerly Tavoy) in Tanintharyi (formerly Tenasserim) by about 400,000 people; its sterotyped characteristic is the mesial ~/I/, found in earlist Bagan inscriptions but by merger there nearly 800 years ago; for further information see Pe Maung Tin (1933) and Okell (1995) [13]. Htawei formerly known as Tavoy is a city of south-eastern Myanmar and is the capital of Tanintharyi Region, formerly known as the Tenasserim is bounded by Mon state to the north, Thailand to the east and south, and the Andaman sea to the west.

Tavoyan retains /-l-/ medial that has since merged into the /-j-/ medial in standard Burmese and can form the following consonant clusters: /gl-/, /kl-/, /kʰl-/, /bl-/, /pl-/, /pʰl-/, /ml-/, /m̥l-/. Examples include မ္လေ (/mlè/ → Standard Burmese /mjè/) for "ground" and ကျောင်း (/kláʊɴ/ → Standard Burmese /tʃáʊɴ/) for "school"[14]. Also, voicing only with unaspirated consonants, whereas in standard Burmese, voicing can occur with both aspirated and unaspirated consonants. Also, there are many loan words from Malay and Thai not found in Standard Burmese. An example is the word for goat, which is hseit (ဆိတ်) in Standard Burmese but be (ဘဲ) in Tavoyan.

In the Tavoyan dialect, terms of endearment, as well as family terms, are considerably different from Standard Burmese. For instance, the terms for "son" and "daughter" are ဖစု (/pʰa̰ òu/) and မိစု (/mḭ òu/) respectively. Moreover, the honorific နောင် (Naung) is used in lieu of မောင် (Maung) for young males.

Another evidence of "Dawei" is "Dhommarazaka" pogoda inscription of Bagan period. It was inscription of Bagan period. It was inscribed in AD 1196 during the region of Bagan King Narapatisithu(AD 1174-1201). In this inscription line 6 to 19, when the demarcation of Bagan is mentioned "Taung-Kar-Htawei" (up to Htawei to the south) and "Taninthaye" (Tanintharyi) are including. Therefore, the name of "Dawei"

appeared particulary since Bagan period, at the time of the first Myanmar Empire (Dawei was established at Myanmar year 1116) is actually meant that the present name Dawei appears as the name of the settlers later and the original name of the city is Tharyarwady, which was established at Myanmar year 1116 according to the saying. As "Dawei" nationality deserves as one nationalist in our country. Actually, Dawei region is a place where local people lived since very ancient Stone Age. After that, Stone Age, Bronze Age and Iron Age culture developed. Moreover, as there has sound evidence of Thargara ancient city, comtemporary to Phu Period, the Dawei people, can be assumed that they are one nationality of high culture in Myanmar.

Dawei(Tavoyan) usage and vocabularies is divided into three main groups. The first one is using Myanmar vocabularies with Dawei speech, the second is the vocabularies same with Myanmar vocabularies and using isolated Dawei words and vocabularies.

In Myanmar word " ထို,ဟို" , " here, there" is used in Dawei "သယ်", "here" and "ဟောက်" , "there". For example "သယ်မျိုး", "ဒီလို" and "ဟောက်မျိုး", "ဟိုလို". The question words in Myanmar "နည်း(သနည်း)", "လဲ(သလဲ)" is used and the Dawei word "လော,လော် " is used instead of "လား(သလား)". For example "ဘာလဲ(what)" and 'ဘာဖြစ်တာလဲ(what happened)" is "ဖြာနူး" and " ဖြာဖြစ်နူး" in Dawei usage. In negative sense of Myanmar word "ဘူး" is not usually used in Dawei word. The negative Dawei word "ဟှ(ရ) " " or "ဟန်း" instead of " No " in Myanmar word. Myanmar adverb word "သိပ်" , "အလွန်", "အလွန်အလွန်" (very , extremely) " is used as Dawei word "ရရာ" , "ရမိရရာ" , "ဗြင်း" . There are many Dawei vocabularies such as "ဝန်းရှင်း " is called in Myanmar " ကိုယ်ဝန်ဆောင်" (pregnant) " , in Dawei word "ကောန်သား" , Myanmar word "ကောင်လေး (boy) " , "Dawei word ဝယ်သား", Myanmar word " ကောင်မလေး (girl) " , Dawei word " ကပ် " , "ပိုက်ဆံ (money) " , Dawei word " ချော့-က်တိုအိုးသီး" , Myanmar word "ကျွဲကောသီး (pomelo)" and Dawei word "သစ်ခတ်ကွား", Myanmar word "ကျားသစ်(leopard)".

The followings are some example parallel sentences of Myanmar (my) and Dawei (dw):

dw : <u>သယ်ဝယ်သား</u> က လှ ဗြင်း ဟှယ် ။
my : <u>ဒီကောင်မလေး</u> က လှ လွန်း တယ် ။
("<u>The girl</u> is so beautiful" in English)

dw : <u>လတ်ဖတ်ရယ်</u> က ရှိ ဗြင်း ဟှယ် ။
my : <u>လက်ဖက်ရည်</u> က ချို လွန်း တယ် ။
("The <u>tea</u> is so sweet" in English)

dw : ကောန်သား <u>ကျောန်း</u> မှန်းမှန် သွား ဟှယ် ။
my : ကောင်လေး <u>ကျောင်း</u> မှန်မှန် တက် တယ် ။
("The boy goes to <u>school</u> regularly" in English)

IV. METHODOLOGY

In this section, we describe the methodology used in the machine translation experiments for this paper.

A. Encoder-Decoder Model

The core idea is to encode a variable-length input sequence of tokens into a sequence of vector representations, and to then decode those representations into a sequence of output tokens. This decoding is conditioned on information from both the latent input vector encodings as well as its own continually updated internal state, motivating the idea that the model should be able to capture meanings and interactions beyond those at the word level. Formally, given source sentence $X= x_1,...,x_n$ and target sentence$Y= y_1,...,y_m$, an NMT system models $p(Y|X)$as a target language sequence model, conditioning the probability of the target word y_t on the target history $Y_{1:t-1}$ and source sentence X . Each x_I and y_t are integer ids given by source and target vocabulary mappings,V_{src} and V_{trg}, built from the training data tokens and represented as one-hot vectors $x_i \in \{0,1\}^{|V_{src}|}$ and $y_t \in \{0,1\}^{|V_{trg}|}$. These are embedded into e-dimensional vector representations, $E_S x_i$ and $E_T y$, using embedding matrices $\mathbb{R}^{e\times|V_{src}|}$ and $E_T \in \mathbb{R}^{e\times|V_{trg}|}$. The target sequence is factorized as $p(Y|X;\theta) = \prod_{t=1}^{m} p(y_t|Y_{1:t-1},X;\theta)$. The model, parameterized by θ, consists of an encoder and a decoder part, which vary depending on the model architecture . $p(y_t|Y_{1:t-1},X;\theta)$ is parameterized via a

softmax output layer over some decoder representation

(1)

where W_o scales to the dimension of the target vocabulary V_{trg}. For parameterizing the encoder and the decoder, Recurrent Neural Networks (RNNs) constitute a natural way of encoding variable-length sequences by updating an internal state and returning a sequence of outputs, where the last output may

$$p(y_t|Y_{1:t-1}, X;) = softmax(W_0 t + b_0)$$

summarize the encoded sequence [15]. However, with longer sequences, a fixed-size vector may not be able to store sufficient information about the sequence, and attention mechanisms help to reduce this burden by dynamically adjusting the representation of the encoded sequence given a state [16].

B. Recurrent Neural Network

The first encoder layer consists of a bi-directional RNN followed by a stack of uni-directional RNNs. Specifically, the first layer produces a forward sequence of hidden states, some non-linear function, such as a Gated Recurrent Unit (GRU) or Long Short Term Memory (LSTM) cell. The reverse RNN processes the source sentence from right to left: and the hidden states from both directions are concatenated. The hidden state h_0 it can incorporate information from both tokens to the left, as well as tokens to the right.

The decoder consists of an RNN to predict one target word at a time through a state vector

$$\mathbf{s}_t = f_{dec}([\mathbf{E}_T \mathbf{y}_{t-1}; \bar{\mathbf{s}}_{t-1}], \mathbf{s}_{t-1}), \tag{2}$$

where f_{dec} is a multi-layer RNN, s_{t-1} the previous state vector, and $\bar{s}_{t-1}$ the source-depend attentional vector. Providing the attentional vector as an input to the first decoder layer is also called input feeding [17]. The initial decoder hidden state is a non-linear transformation of the last encoder hidden state:$s_0 = \tanh(W_{init} h_n + b_{init})$. The attentional vector $\bar{s}_t$ combines the decoder state with a context vector c_t:

$$\bar{\mathbf{s}}_t = \tanh(\mathbf{W}_{\bar{s}}[\mathbf{s}_t; \mathbf{c}_t]), \tag{3}$$

where c_t is a weighted sum of encoder hidden states:$c_t = \sum_{i=1}^{n} \alpha_{ti} h_i$. The attention vector α_t is computed by an attention network $\alpha t_i = \text{softmax}(\text{score}(s_t, h_i))$ where score (s,h) could be defined as

$$\alpha_{ti} = \text{softmax}(\text{score}(\mathbf{s}_t, \mathbf{h}_i))$$
$$\text{score}(\mathbf{s}, \mathbf{h}) = \mathbf{v}_a^\top \tanh(\mathbf{W}_u \mathbf{s} + \mathbf{W}_v \mathbf{h}). \tag{4}$$

, the computation of RNN hidden states cannot be parallelized over time.

C. Transformer

The transformer model [17] uses attention to replace recurrent dependencies, making the representation at time step independent from the other time steps. This allows for parallelization of the computation for all time steps in encoder and decoder.

The embedding is followed by several identical encoder blocks consisting of two core sub layers: self-attention and a feed-forward network. The self-attention mechanism is a variation of the dot-product attention [16] but generalized to three inputs: a query matrix $Q \in \mathbb{R}^{n \times d}$, a key matrix $K \in \mathbb{R}^{n \times d}$, and a value matrix $V \in \mathbb{R}^{n \times d}$, where d denotes the number of hidden units. Vaswani et al. [2017] further extend attention to multiple heads, allowing for focusing on different parts of the input. A single head u produces a context matrix

$$\mathbf{C}_u = \text{softmax}\left(\frac{\mathbf{Q}\mathbf{W}_u^Q (\mathbf{K}\mathbf{W}_u^K)^T}{\sqrt{d_u}}\right)\mathbf{V}\mathbf{W}_u^V, \tag{5}$$

where matrices W_u^Q, W_u^K b and W_u^V belong to $R^{d \times du}$. The final context matrix is given by concatenating the heads, followed by a linear transformation : $C = [C_1; ...; C_h]W^O$. The number of hidden units chosen to be a multiple of the number of units per head. Given a sequence of hidden states h_i (or input embeddings), concatenated to $H \in \mathbb{R}^{n \times d}$, the encoder computes self-attention using Q=K=V=H. The second sub network of an encoder block is a feed-forward network with ReLU activation defined as

$$FFN(\mathbf{x}) = \max(0, \mathbf{x}\mathbf{W}_1 + \mathbf{b}_1)\mathbf{W}_2 + \mathbf{b}_2, \tag{6}$$

which again is easily parallelizable across time steps. Each sub layer, self-attention and feed-forward network, is followed by a post-processing stack of dropout, layer normalization [18], and residual connection. A complete encoder block hence consists of Self-attention → Post-process → Feed-forward → Post-process can be stacked to form a multi-layer encoder. To maintain auto-regressiveness of the model, attention on future time steps is masked out accordingly [17]. In addition to self-attention, a source attention layer which uses the encoder hidden

states as key and value inputs is added. The sequence of operations of a single decoder block is:

Selfattention→Postprocess→Encoderattention→Post -process→Feed-forward→Post-process

Multiple blocks are stacked to form the full decoder network and the representation of the last block is fed into the output layer of Equation.

V. EXPERIMENTS

A.Corpus Statistics

We used 9K Myanmar sentences (without name entity tags) of the ASEAN-MT Parallel Corpus [19], which is a parallel corpus in the travel domain. It contains six main categories and they are people (greeting, introduction and communication), survival (transportation, accommodation and finance), food (food, beverage and restaurant), fun (recreation, traveling, shopping and nightlife), resource (number, time and accuracy), special needs (emergency and health). We used 9k Myanmar-Dawei parallel corpus and 6,883 sentences for traning, 1,215 sentences for development and 902 sentences for testing.

B. Syllable Segmentation

We defined Generally, Myanmar words are composed of multiple syllables, and most of the syllables are composed of more than one character. Syllables are composed of Myanmar words. If we only focus on consonant-based syllables, the structure of the syllable can be described with Backus normal form (BNF) as follows:

Syllable := CMW[CK][D]

Here, C stands for consonants, M for medials, V for vowel, K for vowel killer character, and D for diacritic characters. Myanmar syllable segmentation can be done with a rule-based approach, finite state automation (FSA) or regular expressions (RE)

(https://github.com/yekyawthu/sylbreak).

In our experiments, we used RE based Myanmar syllable segmentation tool named. The following is an example of syllable segmentation for a Dawei sentence in our corpus and the meaning is

Unsegmented Dawei sentence:

dw: ဘယ်နားက လာဟို့ ဘယ်နားသွားဟို့ ဘဲ့သေးနူး ။

Syllable Segmented Dawei sentence:

dw: ဘယ် နား က လာ ဟို့ ဘယ် နား သွား ဟို့ ဘဲ့ သေး နူး ။

C. Byte-Pair-Encoding

Sennrich et al., 2016 [21] proposed a method to enable open-vocabulary translation of rare and unknown words as a sequence of subword units as a sequence of subword units representing BPE algorithm [22]. The input is a monolingual corpus for a language (one side of the parallel training data, in our case) and starts with an initial vocabulary, the characters in the text corpus. The vocabulary is updated using an iterative greedy algorithm. In every iteration, the most frequent bigram (based on the currebt vocabulary) in the corpus is added to the vocabulary (the merge operation). The corpus is again encoded the updated vocabulary, and this process is repeated for a predetermined number of merge operations. The number of merge operations is the only hyper-parameter of the system that needs to be tuned. A new word can be segmented by looking up the learnt vocabulary.

D. Moses SMT System

We used the Moses toolkit (Koehn et al., 2007) for training the operation sequence model (OSM) statistical machine translation systems. We did not consider phrase-based statistical machine translation (PBSMT) and hierarchical phrase-based statistical machine translation (HPBSMT), because the OSM approach achieved the highest BLEU [7] and RIBES [8] scores among three approaches [24] for both Myanmar-Dawei to Dawei-Myanmar statistical machine translations. The word-segmented (i.e., Syllable-BPE) source language was aligned with the word-segmented target language using GIZA++. The alignment was symmetrized by grow-diag-final and heuristic. The lexicalized reordering model was trained with the msd-bidirectional-fe option. We used KenLM [9] for training the 5-gram language model with modified Kneser- Ney discounting. Minimum error rate training (MERT) was used to tune the decoder parameters, and the decoding was done using the Moses decoder (version 2.1.1) [23]. We used the default settings of Moses for all experiments.

E. Framework of NMT System

An open-source sequence-to-sequence toolkit for NMT written in Python [20] and built on Apache MXNET, the toolkit offers scalable training and

inference for the three most prominent encoder - decoder architectures: attentional recurrent neural network, self-attentional transformers.

F. Training Details

We used the Sockeye toolkit, which is based on MXNet, to train NMT models. The initial learning rate is set to 0.0001. All experiments are runned with maximum epoch. If the performance on the validation set has improves for 8 checkpoints, the learning rate is multiplied by 8 checkpoints. All the neural networks have eight layers. For RNN Seq2Seq, the encoder has one bi-directional LSTM and six stacked unidirectional LSTMs, and the encoder is a stack of eight unidirectional LSTMs. The size of hidden states is 512. We apply layer-normalization and label smoothing (0.1) in all models. We tie the source and target embedding. The dropout rate of the embedding and transformer blocks is set to (0.1). The dropout rate of RNNs is (0.2). The attention mechanism in the transformer has eight heads.

We applied three different batch sizes (128, 256 ,512 and 1024) for RNN and Transformer architectures. The learning rates varies from 0.0001 to 0.0005. Two memory cell types GRU and LSTM were used for the RNN and transformer. The comparison was done for both SMT (i.e., PBSMT, HPBSMT) and NMT (RNN, Transformer) techniques. All experiments are run on NVIDIA Tesla K80 24GB GDDR5, We trained all models for the maximum number of epochs using the AdaGrad and adaptive moment estimation (Adam) optimizer. The BPE segmentation models were trained with a vocabulary of 8,000.

VI. Results and Discussion

In discussion of Table I, different batch sizes gained comparable results for both RNN and Transformer architecture for both cell types. The results of Table II, Batch size 128 is the best score in RNN architecture. Therefore, we are runned batch size 128 with two different cell types LSTM and GRU with different learning rates. In evidence of the results of Table III, the learning rate 0.0005 is the best score of bi-directional Myanmar-Dawei machine translation. In comparing the results of two cell type of GRU and LSTM, LSTM is more suitable than GRU in RNN architecture. For the results of Table III and Table IV, we can be clearly say that RNN is better results than transformer model architecture.

Discussion about batch size 256 shown in table V and table VI. LSTM cell type of RNN architecture is better results than GRU in bi-directional Myanmar to Dawei machine translation for the result of table V. But in transformer architecture of batch size 256 GRU cell type is get better results than LSTM shown in the results of table VI.

In Table V, the learning rate 0.0005 is gained the highest score the Myanmar-Dawei machine translation that score is 44.68 on batch size 256 on LSTM cell type in RNN architecture and Dawei-Myanmar machine translation highest score is 61.84 is on batch size 128 on LSTM type on RNN. In summarization of above facts, LSTM cell type of RNN architecture is best score for Myanmar to Dawei bi-directional machine translation.

In summarization of the results of batch size 512 on Table VII and Table VIII LSTM cell type is best score for Myanmar to Dawei machine translation and GRU cell type is best score for Dawei to Myanmar machine translation. Comparing the discussion results is shown in Table IX. We found that LSTM cell type with RNN architecture is the best for Dawei-Myanmar bi-directional machine translation. Myanmar to Dawei NMT achieved comparable results with PBSMT and HPBSMT. Moreover, Dawei to Myanmar RNN machine translation performance achieved higher BLEU scores than PBSMT (+1.06 BLEU) and HPBSMT (+1.37 BLEU) even with the limited parallel corpus.

TABLE I. BLEU SCORES OF SYLLABLE-BPE SEGMENTATION FOR VARIOUS BATCH SIZES WITH TWO DIFFERENT CELL TYPE FOR TRANSFORMER MODEL

Batch size	Transformer			
	LSTM		*GRU*	
	my-dw	*dw-my*	*my-dw*	*dw-my*
128	42.32	57.5	42.31	57.50
256	42.16	57.57	**42.37**	**57.84**
512	**42.21**	**57.6**	42.21	57.60
1024	41.85	57.12	41.85	56.38

TABLE II. BLEU SCORES OF SYLLABLE-BPE SEGMENTATION FOR VARIOUS BATCH SIZES WITH TWO DIFFERENT CELL TYPE FOR RNN MODEL

Batch size	RNN			
	LSTM		*GRU*	
	my-dw	*dw-my*	*my-dw*	*dw-my*
128	**40.96**	**57.51**	**41.29**	**57.43**
256	39.5	54.13	40.67	56.63
512	38.86	55.95	38.95	52.48
1024	35.82	51.08	37.43	53.63

TABLE III. BLEU SCORES OF SYLLABLE-BPE SEGMENTATION FOR VARIOUS LEARNING RATES ON DIFFERENT CELL TYPE FOR RNN MODEL ON BATCH 128

Learning	RNN

rate	LSTM		GRU	
	my-dw	dw-my	my-dw	dw-my
0.0002	43.47	58.04	41.6	59.27
0.0003	43.39	60.03	43.22	59.39
0.0004	42.94	60.6	42.94	60.00
0.0005	**44.24**	**61.84**	**43.23**	**60.38**

TABLE IV. BLEU SCORES OF SYLLABLE-BPE SEGMENTATION FOR VARIOUS LEARNING RATES ON DIFFERENT CELL TYPE FOR TRANSFORMER MODEL ON BATCH 128

Learning rate	transformer			
	LSTM		GRU	
	my-dw	dw-my	my-dw	dw-my
0.0002	41.97	58.72	**42.69**	58.49
0.0003	42.62	58.84	42.23	60.17
0.0004	42.72	**59.09**	42.50	59.94
0.0005	**43.03**	58.85	41.84	**60.44**

TABLE V. BLEU SCORES OF SYLLABLE-BPE SEGMENTATION FOR VARIOUS LEARNING RATES ON DIFFERENT CELL TYPE FOR RNN MODEL ON BATCH 256

Learning rate	RNN			
	LSTM		GRU	
	my-dw	dw-my	my-dw	dw-my
0.0002	41.93	59.55	43.34	59.46
0.0003	42.68	59.95	43.79	60.61
0.0004	43.28	59.55	**43.84**	60.77
0.0005	**44.68**	**61.31**	43.14	**61.24**

TABLE VI. BLEU SCORES OF SYLLABLE-BPE SEGMENTATION FOR VARIOUS LEARNING RATES ON DIFFERENT CELL TYPE FOR TRANSFORMER MODEL ON BATCH 256

Learning rate	transformer			
	LSTM		GRU	
	my-dw	dw-my	my-dw	dw-my
0.0002	42.84	58.25	42.32	57.53
0.0003	42.57	58.35	43.79	**58.35**
0.0004	42.48	58.61	**43.84**	59.17
0.0005	**42.70**	**59.17**	43.14	59.17

TABLE VII. BLEU SCORES OF SYLLABLE-BPE SEGMENTATION FOR VARIOUS LEARNING RATES ON DIFFERENT CELL TYPE FOR RNN MODEL ON BATCH 512

Learning rate	RNN			
	LSTM		GRU	
	my-dw	dw-my	my-dw	dw-my
0.0002	42.29	57.27	41.50	57.99
0.0003	43.46	59.64	42.71	57.19
0.0004	**45.58**	59.23	43.37	**60.04**
0.0005	43.72	**59.86**	**44.23**	60.20

TABLE VIII. BLEU SCORES OF SYLLABLE-BPE SEGMENTATION FOR VARIOUS LEARNING RATES ON DIFFERENT CELL TYPE FOR TRANSFORMER MODEL ON BATCH 512

Learning rate	transformer			
	LSTM		GRU	
	my-dw	dw-my	my-dw	dw-my
0.0002	41.83	57.84	**42.50**	57.88
0.0003	**42.97**	58.41	42.04	58.41
0.0004	42.12	58.82	42.12	57.93
0.0005	41.22	**59.71**	41.94	**60.44**

TABLE IX. BLEU SCORES OF COMPARISON OF SMT AND NMT

	PBSMT	HPBSMT	RNN	Trasnsformer
my-dw	44.80	45.44	44.68	43.03
dw-my	60.78	60.47	**61.84**	60.44

VII. CONCLUSION

In this section, This work is the first evaluation of the quality of neural machine translation between Myanmar (Burmese) and Dawei (Tavoyan). We are developing a Myanmar-Dawei parallel corpus (around 9K sentences) based on the Myanmar language of ASEAN MT corpus and implemented on two prominent neural machine translation system : Recurrent Neural Network (RNN) and Transformer with syllable segmentation. We also investigated various hyper-parameters such as batch size, learning rate and cell types (GRU and LSTM). We proved that LSTM cell type with RNN architecture is the best for Dawei-Myanmar bi-directional machine translation. We are compared results between SMT (PBSMT and HPSMT) and NMT (RNN) in Myanmar to Dawei machine translation. We found that LSTM cell type with RNN architecture is the best for Dawei-Myanmar bi-directional machine translation. Myanmar to Dawei NMT achieved comparable results with PBSMT and HPBSMT. Moreover, Dawei to Myanmar RNN machine translation performance achieved higher BLEU scores than PBSMT (+1.06 BLEU) and HPBSMT (+1.37 BLEU) even with the limited parallel corpus. In the near future, we plan to conduct a further study with a focus on NMT models with one more subword segmentation scheme SentencePiece for Myanmar-Dawei NMT. Moreover, we intend to investigate SMT and NMT approaches for Myeik dialectal languages.

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Preprocessing of YouTube Myanmar Music Comments for Sentiment Analysis

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Abstract

Over the years, social media sites have become an important role of communication and a source of the tremendous amount of real-time data of images, videos, etc. YouTube is likely the most popular of them, with millions of uploaded videos and billions of comments for all these videos. This paper presents a new Music dataset, which has the YouTube comments written in Myanmar language, to be applied in sentiment analysis. Data preprocessing is very important for our language because it allows improving the quality of the raw data and converting the raw data into a clean dataset. The preprocessing of music comments is followed by basic phase, removing phase, segmentation phase, replacement phase and translation phase. The outcome of YouTube comment preprocessing will aid in better sentiment analysis. Results show that the preprocessing approaches give a significant effect on the musical opinion extraction process using information gain.

Keywords: *preprocessing, opinion extraction, Myanmar language, YouTube comments, information gain*

I. INTRODUCTION

Sentiment analysis is also known as opinion mining, which gives positive, negative, neutral feedback from users. This is widely used today in selecting a restaurant, in buying any product from the e-commerce site, feedback of any hotel [1]. YouTube is one of the comprehensive video information sources on the web where the video is uploaded continuously in real-time. It is one of the most popular sites in social media, where users interact with commenting, sharing and rating videos [2]. The main work of the paper is to retrieve and preprocess the YouTube comments written in Myanmar language from the Music domain. Data pre-processing is the process of cleaning and preparing the data for analysis. Online comments normally contain lots of noise. In addition, many words in the comments do not have an effect on the general orientation of it. Remaining those words makes the dimensionality of the problem high and hence the sentiment analysis more difficult since each word in the comments is treated as one dimension. Here is the hypothesis of having the data properly pre-processed: to cut the noise in the comment should help improve the performance of the classifier and speed up the classification process, thus aiding in real time sentiment analysis [3]. In this paper, the preprocessing of the data is discussed in detail. Most of the work on sentiment analysis has targeted English text since a language like the Myanmar language which holds the 29th place among the most 30 languages spoken worldwide [4] did not get enough focus. The main aim of this study was to check the effect of several pre-processing approaches in extracting musical opinions by using information gain.

The rest of this article is organized as follows. Section 2 reviews related work. Section 3 reports data collection. Section 4 explains preprocessing steps in detail. Information gain in preprocessed data for musical opinion extraction is explained in section 5. The experimental results are described and analyzed in section 6. Finally, the conclusion and future work are explained.

II. RELATED WORK

This section presents earlier work that deals with preprocessing data for sentiment analysis.

R. Duwairi and M. El-Orfali [5] solved the problem of three perspectives of Arabic sentiment analysis: representation perspective, classifier perspective and dataset perspective. In representation perspective, they described the reviews in seven representations as a pre-classification task. They are base-line vectors, stemmed vectors and so on. In

classifier perspective, they used three supervised learning methods (K-NN, SVM and Naïve Bayes) to classify the reviews on Arabic language into positive or negative classes. In dataset perspective, they analyzed the impacts of the data properties on their obtained results. Politics dataset and Movie dataset were applied for the experiments.

E. Haddia, X. Liua and Y. Shib [3] investigated the text pre-processing role in sentiment analysis, and reported on experimental results that demonstrated that with suitable feature selection and representation, the accuracies of sentiment analysis using Support Vector Machines (SVM) may be remarkably improved. The whole preprocessing steps involves text cleaning, whitespace removing, abbreviation expanding, negation handling, stemming, stopwords removing and at last feature selection. They described extensive experimental results with suitable text pre-processing methods containing data transformation and filtering for enhancing the classifier performance.

The authors [6] focused on the preprocessing techniques on Twitter data to perform sentiment analysis. Their preprocessing steps include filtering and the removing of URLs, questions, special characters and retweets. They applied SentiWordNet for analysis, bigrams for increasing the accuracy of the classifier and used SES algorithm to calculate sentiment.

The researchers [7] explained the required information to get preprocessed reviews for searching sentiment. They used the StringToWordVector filter in order to do the preprocessing task in WEKA. The filer included weighting scheme (TF-IDF), stop-words removal (Rainbow list), stemming (Snowball stemmer), tokenization (NGram Tokenizer) and feature selection. For the experiments, they used three freely available twitter dataset and four classifier namely Support Vector Machine, Naïve Bayes, k-Nearest Neighbor and C4.5.

The authors [8] explored the impact on the pre-processing strategies in online movie reviews sentiment analysis and used supervised machine learning approach, linear and non-linear kernel support vector machine (SVM), to classify the reviews. Three tier of cleaning strategies are described and they were remove stop word (tier-1), remove stop word + meaningless words (tier-2)and remove stop word + meaningless words + numbers + words less than 3 character (tier-3). Their important finding was TF and TFIDF are achieved the best results of features representation on the SVM with non-linear kernel.

III. Data Collection

Before preprocessing on the data, we need to collect the data from the source. This system creates Music comment dataset. The dataset has been prepared manually by collecting original unprocessed comments from YouTube. The comments from YouTube music video are scraped using YouTube Comment Scraper. More than 32,000 comments from different music's web pages are gathered and have been evaluated by two evaluators. Only about 14 percent of comments are not having any noise and for this reason any cleaning is not required.

IV. Data Preprocessing

After collecting the data, it is ready for the preprocessing. Preprocessing step is one of the basic tasks in sentiment analysis and it will clean the dataset by diminishing its complexity to make ready the data for the opinion extraction task. There are several stages in the preprocessing: from simple data cleaning by removing URLs, white spaces and stopwords up to more advanced normalization techniques such as Myanmar word segmentation.

Generally, the main purpose of the preprocessing is to keep all the Myanmar words that are valuable for the analysis and, at the same moment, get rid of all others, and the dataset should be kept in one uniform format. The overall work of the system is shown in Figure 1.

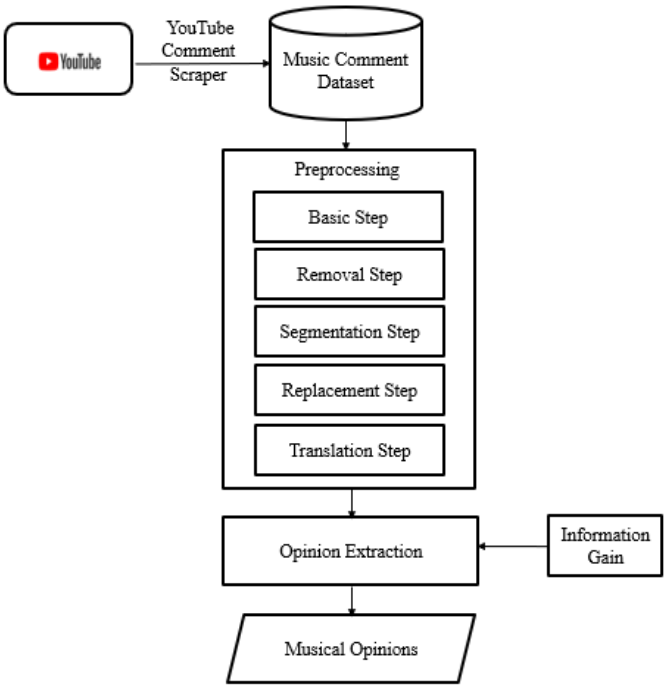

Figure 1. Overall Work

The data preprocessing can vary based on the source of the data. Key steps of comments preprocessing are listed below with explanations: In this task, five steps of cleaning approaches were developed, which called basic step, removal step, segmentation step, replacement step and translation step. Some steps are done manually, and some are solved by our created rules because it is impossible to use other popular tools for our language.

A. Basic Step

This is the first step of preprocessing and the steps are shown in Table 1. Zawgyi comments are converted to Unicode using a conversion tool and the spellings are corrected manually.

- Font Mapping
  Most users use Zawgyi font in the comments and can write with many forms.
  For example: ကြိုက် ကြိုက်
  This problem is addressed by converting Zawgyi to Unicode with online font converter [9] .
- Spelling Correction
  Human mistake is not supposed to have an effect on how computers learn, and that's essential. If computers do the same errors as humans make, then they'll be as much unusable as those humans who make errors often. There is no spelling checker for our language and we do this works manually.
  For example: သခ်ၤင္း, သီခ်င္ for the word "song"
  ကို္ကၠ် for the word "like"

TABLE I. BASIC STEPS

No	Basic Steps	How to address
1	Font Mapping	Tool
2	Spelling Correction	Manual

B. Removal Step

In this step, unnecessary words or comments will be removed since they do not play an important part in deciding the sentiment.

- Whitespace and Blank Lines Removal
  When dealing with raw comments, we normally have a set of words, including many items we are not interested in, such as blank lines, line breaks and whitespace. Blank lines are also unnecessary to calculate the real number of comments.
- URLs, Hyperlinks and Hashtags Removal
  URLs, Hyperlinks and Hashtags can be found commonly in the comments, and these are unnecessary items in the dataset.
- Punctuations and Non-Myanmar Words Removal
  To remove the noise from the comments, punctuations such as (!, -, *, &) and other non-Myanmar words such as Thai words, Rakhine words in the comments are deleted. The special characters may be combined with the words and those words were inaccessible in the dictionary.
- Singers' Names Removal
  The way of removing singers' names is to have a predetermined list of names, which can be deleted from the comments.
- Rare Words Removal
  Some words appeared five or a smaller number of times from the comments and these rare words are removed. About 36.8% of the words occurred just once.
- Repeated Syllables Removal
  Sometimes there is a need to make some corrections for the quality of the dataset. This mentions to YouTube, where emotions can play a title role and the comments contain multiple syllables.
  For example: "ခ်စ္စစ္စစ္စ္" (looooove) instead of "ခ်စ္" (love).
  Repeated Myanmar syllables are reduced and converted to the original word.
- Duplicate Comments Removal
  It is noticed that multiple duplicates of the same writer are very common in the comments. The writer duplicated the same comments in many sentences. In many instances, these duplicates come from errors in data analysis process and it is suggested to remove them in order to reduce bias in our analysis.
- Stopwords Removal
  Stopwords are very common in the comments, and they're so repeated, that their semantic meaning is lost. Words like "တယ္၊ သည္" are some examples of stopwords. Irrelevant and frequent words consequently seem to be nothing but very intrusive noise. Some stopwords are shown in Table 2 and removing steps are shown in Table 3.

TABLE II. SAMPLE OF STOPWORDS

Stopwords		
က	တွေ	နော်
ကတော့	တာလဲ	နေ
ကလည်း	တာပါ	ပါ
ကွာ	တော့	ပါရစေ
ကွယ်	တယ်	ပါစေ
ကို	တေ	ပါတယ်
ကြောင့်	တွေ့	ပြီး
ချက်	တယ်နော်	ပီး
စေ	တာ	ရှင်
ဆိုတဲ့	တဲ့	တုန်း

TABLE III. REMOVING STEPS

No	Removing Steps	How to address
1	Whitespace and Blank Lines Removal	Manual
2	URLs, Hyperlinks and Hashtags Removal	Regular Expression
3	Punctuations and Non-Myanmar Words Removal	Regular Expression and Manual
4	Singers' Names Removal	Collected Name List
5	Rare Words Removal	Count Frequency
6	Repeated Syllables Removal	Regular Expression
7	Duplicate Comments Removal	Manual
8	Stopwords Removal	Collected Stopwords List

C. Segmentation Step

After removing the unnecessary words, words in the comments are needed to be segmented for better handling.

- Word Segmentation
  Word segmentation is done using Myanmar Word Segmentation Version 1.0 of Natural Language Processing Lab of University of Computer Studies, Yangon (UCSY) [10]. Some words are recombined after word segmentation to have opinion words. Some of merge rules are shown in Table 4 and segmentation step is shown in Table 5.

TABLE IV. SAMPLE OF MERGE RULES

Words	Merge
ကွန်_ ကရက်_ ကျူ_ လေး_ ရှင်း	ကွန်ကရက်ကျူလေးရှင်း
အ_ လန်း_ ဇယား	အလန်းဇယား
ရင်ထဲ_ ထိ	ရင်ထဲထိ
မ_ ကြိုက်	မကြိုက်
ဘို_ ကေ	ဘိုကေ
မ_ မိုက်	မမိုက်
ငှက်ဆိုး_ သံ	ငှက်ဆိုးသံ
ပ_ ရို_ သံ_ မ_ ပေါက်	ပရိုသံမပေါက်
ဝင်း_ နား	ဝင်းနား
နားဝင်_ ဆိုး	နားဝင်ဆိုး

TABLE V. SEGMENTATION STEP

No	Segmentation Step	How to address
1	Word Segmentation	Tool

D. Replacement Step

In this step, some words in the comments will be replaced with other words. The replacement words are important to define the sentiment words.

- Replacement of Modifier Synonym
  Certain Myanmar synonym adverbs like အားကြီး၊ အကုန်၊ အသေ are replaced by အရမ်း (very).
- Root Words Replacement
  Words in comparative form are replaced with the basic form.
  Examples of such words are ပိုကြိုက် (like more), အကြိုက်ဆုံး (like most). These words are converted to the basic form ကြိုက် (like).
  This is done by using our created root word rules and some rules are shown in Table 6 and replacement steps are shown in Table 7.

TABLE VI. STEMMING RULES

Words	Stem
အချစ်, ချစ်စရာ, ချစ်သူ ချစ်ဖို့ကောင်း, ချစ်စရာလေး	ချစ်
ရှူးသွပ်, အရှူး, စောက်ရှူး ရှူးကြောင်ကြောင်, အရှူးထ	ရှူး
သတ်ပစ်ချင်, သတ်သတ်, သတ်ပလိုက်	သတ်
ကီးကြောင်ကြီး, ကီးကမမှန် ကီးကလဲကြောင်, ကီးမမှန်	ကီးကြောင်
အကြိုက်, အကြိုက်ဆုံး, ကြိုက်ဆုံး	ကြိုက်

- Emoticons Replacement
  In this busy world, people do not bother likes to use a large amounts of sentences for the posts. But they can simply express a huge sentence using a single emoticon. It may cause the entire polarity of a comment. Emoticons, which are composed of non-alphabets also play a role in sentiment analysis. ":), :(, :D, xD", all these, when handled precisely, can assist with a better sentiment analysis. They are replaced with their respective Myanmar words.
  For example: :) = ပြုံး (smile) :(= မဲ့ (sad)

TABLE VII. REPLACEMENT STEPS

No	Replacement Steps	How to address
1	Replacement of Modifier Synonym	Collected Synonyms
2	Root Word Replacement	Rule-based
3	Emoticons Replacement	Rules-based

E. Translation Step

This final step is valuable in normalizing the comments. This is not always required to use and it relies on the need of a problem. In this step, some words are translated to Myanmar words using collected rules. Table 8 depicts the translation steps.

- Acronyms Translation
  Some human expressions, like "lol, wow, haha" can also be a helpful information when working on sentiment analysis. English acronyms are changed to similar Myanmar words.
  For example: 'haha' to ရယ်
- Combined Words Translation
  English-Myanmar combined words are converted to meaningful Myanmar words.
  For example: 'ဒေကာင်း' , 'ဒကောင်း' to အရမ်းကောင်း (very good).
- Pronunciation Words Translation
  Myanmar words with English pronunciation are changed to Myanmar words.
  For example: ဂွတ် (good in English) to ကောင်း
- Loan Words Translation
  Several useful English words and loan words are first converted to lower case and then translated to similar Myanmar words [11].
  For example: 'good' to ကောင်း, 'like' to ကြိုက်

TABLE VIII. TRANSLATION STEPS

No	Translation Steps	How to address
1	Acronyms Translation	Rule-based
2	Combined words Translation	Rule-based
3	Pronunciation words Translation	Rule-based
4	Loan words Translation	Rule-based

V. INFORMATION GAIN IN PREPROCESSED DATA FOR MUSICAL OPINIONS EXTRACTION

Properly identified comments present a baseline of information as an input to different systems. Different algorithms can be applied on the proposed dataset. Opinion extraction is the process of identifying and removing as much of the irrelevant and redundant information as possible. The Information Gain (IG) is applied to the dataset as an opinion extraction method. It computes how mixed up the words are [12] and is used to calculate the pertinence of attribute A in class C (positive or negative) in the domain for sentiment analysis. The higher the mutual information value between classes C and attribute A, the higher the pertinence between classes and attribute.

The schemes used for opinions extraction in IG include entropy of class, conditional entropy and information gain. These are based on the following values:

C: a class either positive (Pos) or negative (Neg)
A: attributes
P (C): probability of class
H (C |A): conditional probability of the class in a given attribute
I (C | A): information gain for of the class in a given attribute

A. Entropy of Class, H(C)

The entropy of the class is defined to be:

$$\mathrm{H}(C) = -\Sigma_{cEC}\, p(C) \log p(C) \qquad (1)$$
$$= -\mathrm{p}(\mathrm{Pos}) \log p(\mathrm{Pos}) - p(\mathrm{Neg}) \log p(\mathrm{Neg})$$

Assume we have balance comments, the probability of positive and negative class C is equivalent to 0.5. As a result, the entropy of classes, H(C) is 1.

B. Conditional Entropy, H (C | A)

The conditional entropy of the classis defined to be:

$$\mathrm{H}(C \mid A) = -\Sigma_{cEC}\, p(C \mid A) \log p(C \mid A) \qquad (2)$$

The following is the example of how to calculate the conditional entropy of attribute (**သီခြင်း**) and it is needed to know the count of this attribute in positive comments and negative comments.

$$\mathrm{H}(C \mid \text{သီချင်း}) = -\Sigma_{cEC}\, \mathrm{p}(\mathrm{Pos} \mid \text{သီချင်း}) \log p(\mathrm{Neg} \mid \text{သီချင်း})$$
$$= -\mathrm{p}(\mathrm{Pos} \mid \text{သီချင်း}) \log p(\mathrm{Pos} \mid \text{သီချင်း}) -$$
$$\mathrm{p}(\mathrm{Neg} \mid \text{သီချင်း}) \log p(\mathrm{Neg} \mid \text{သီချင်း})$$

C. Information Gain, I (C | A)

Information gain is defined to be:

$$I(C \mid A) = \mathrm{H}(C) - H(C \mid A) \quad (3)$$

The minimum value of I $(C \mid A)$ occurs if only if H $(C \mid A)$ =1 which means classes and attribute are not correlated at all. In contrast, we tend to select attribute A that mainly appears in one class C either positive or negative. This means that the best opinions are the set of attributes that only occur in one class[13]. Given the training dataset, we computed the information gain for words and removed from the feature space those terms whose information gain was less than some predetermined threshold. The calculation includes the estimation of the conditional probabilities of a category given a term, and entropy computations [14]. The procedure for opinion extraction from the dataset is shown in Fig. 2.

Input: preprocessed comments from dataset

Output: positive and negative musical opinions

for each word A from preprocessed comments do

    calculate H (C|A)

    calculate I (C|A)

end for

for I (C|A) of each word A do

    d= difference between two classes of I (C|A)

    if (d < threshold) then

        A is not selected as opinion

    else if (I (Pos|A) > I (Neg|A)) then

        pos_opinion+=A

    else neg_opinion+=A

    end if

end for

Figure 2. Opinion Extraction Procedure

Some words commonly occur in positive class or negative class only and they are selected as opinions. Unfortunately, it may occur only a few times since the sentiment can be expressed in a various way. As a result, overfitting occurs since those words do not appear. Some always appear in both classes and those words are not selected as musical opinions because it cannot differentiate the class to which it belongs.

VI. Performance Evaluation

Among 32000 comments, about 14% of the comments are not needed to be preprocessed exception in font mapping step. After applying each preprocessing approaches separately, and a combination of them, the comments that are preprocessed become an input for machine learning algorithms that are utilized for opinion extraction and sentiment analysis.

This research work studies that text preprocessing impacts the accuracy of opinion extraction in case of information gain. The accuracy is more increased in opinion extraction because of the replacement and translation steps. The performance of the proposed system is evaluated with our preprocessed data (Section 4) and for this purpose five experiments are done to extract positive and negative musical opinions at different threshold values, and these are shown in Table 9 w.r.t the accuracy rate % and error rate %.

TABLE IX. Five Experiments in Musical Opinions Extraction

Exp. no	Thres-hold value	No. of musical opinions		Accuracy rate %		Error rate %	
		pos	*neg*	*pos*	*neg*	*pos*	*neg*
1	0.01	248	302	76%	81%	24%	19%
2	0.03	226	289	79%	85%	21%	15%
3	0.05	211	253	83%	88%	17%	12%
4	0.07	204	235	87%	74%	13%	26%
5	0.09	185	223	81%	87%	19%	13%

From the above table, it is shown that the best values of the performance measures are for that experiment with threshold value equals 0.07 for positive words and 0.05 for negative words. The most important words are reduced to 204 for positive opinions and 253 for negative opinions. Fig. 3 shows the number of opinions resulted after applying the five experiments at the above threshold values.

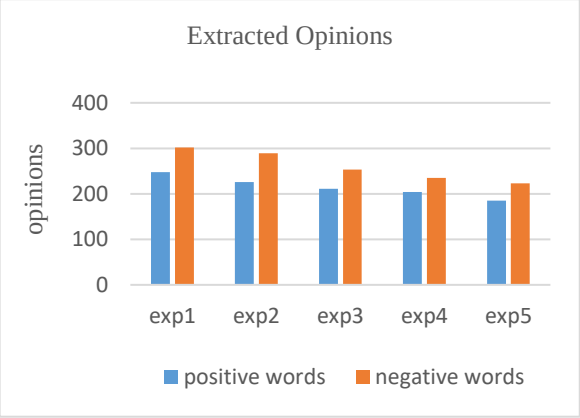

Figure 3. No. of Opinions for Five Experiments

VII. CONCLUSION

The dataset is called the Music dataset and it consists of about 32,000 comments. These comments were collected by the authors of this paper from YouTube website. The comments were preprocessed in several ways and the effects of these are important for accuracy of sentiment analysis. Opinions are extracted using information gain and these are evaluated with five experiments. We applied preprocessing approaches and investigated their impact on accuracy for opinion extraction using information gain.

This research work has opened many places for future direction. Other big challenges of Myanmar sentiment analysis are also needed to be addressed such as sarcasm detection, slang detection, negation handling, spam and fake opinion detection and so on.

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Sentence-Final Prosody Analysis of Japanese Communicative Speech Based on the Command-Response Model

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Abstract

Aiming at communicative speech synthesis, we analyzed sentence-final prosody characteristics through subjective impression on constituting lexicons. Since Japanese sentence-final particles and postpositionals are expected to be employed to generate communicative prosody showing speaker's intention and attitudes, we designed 52 single-phrase utterances showing different strength of the speaker's impressions about judgment. These impressions were quantified in Semantic Differential (SD) scales. F0 contour characteristics were analyzed by using the command-response model. To cope with sentence final F0 characteristics, an additional accent command was introduced for F0 rise and drop of sentence-final particles. The analysis showed systematic communicative prosody control by the accent command reflecting effect of judgment impressions which can be obtained from constituting lexicons. These results indicate possibility of sentence-final prosody control using impression obtained from lexicons constituting output sentences.

Keywords— *speech synthesis, communicative speech, Semantic Differential (SD), Command-Response Model, linguistic modality*

I. INTRODUCTION

Prosodies in communicative speech have wider variations than those in reading-style speech. In speech engineering field, some of them have been studied as para-linguistic or expressive speech prosody including emotional one [1,2,3,4]. Most of these studies have focused on predetermined speech categories such as emotional ones for speech analysis and synthesis. In real-field communications, there exist much wider variations which cannot be treated by pre-determined variation categories, which requires detailed analyses between constituting lexicons and communicative prosody [5].

Recently, communicative prosody variations have been analyzed for Japanese human interactions from data analysis viewpoint and sentence-final prosodic characteristics have been analyzed [6,7]. These sentence-final prosody variations in Japanese speech have already been studied by phoneticians relating to their linguistic attributes [8]. These studies in speech engineering and phonetics support the control possibilities of communicative prosody using lexicons constituting sentence-final parts.

For sentence-final prosody, we have analyzed single phrase utterances consisting of a verb, postpositionals working as a modality of judgment, and final particles working as a modality of utterance [9] as shown in the top row (phrase form) of Table I. Throughout this analysis, we could have found the correlation between sentence-final lexicons and F0 drops in the final mora.

In this paper, we have analyzed sentence-final prosody of these single phrase utterances using the command-response model [10] to understand the prosody control characteristics more clearly and to confirm quantitative control possibilities from constituting lexicons. In the following sections, in Section II we introduce our previous studies on the correspondence between sentence-final prosody in communicative speech and impressions obtained from constituting particles and postpositionals. Section III describes the analysis method we employed; the introduction of the command-response model (what is called Fujisaki Model), newly introduced parameters for sentence-final F0 control, and the impression measurement criteria. Section IV describes the experimental results of model parameter characteristics in communicative speech. Finally, we sum up the findings in Section V.

II. BACKGROUNG

Aiming at communicative speech synthesis, we have been studying the differences between communicative prosody and reading-style one based on constituting lexicons of the utterance [11,12,13]. Through these analyses, we have found strong correlation between F0 height and the strength of degree adverbs [11]. Furthermore, based on the correlations between F0 shape and impressions found in short utterances [12], we have shown a possibility of communicative speech computation using multi-dimensional impressions obtained from constituting lexicons [13].

In the most recent study, we found the correlation between Japanese sentence-final communicative prosody and impressions given by constituting lexicons [9]. In Japanese sentences, speaker's subjective information what is called linguistic modality appears in grammar structure. Masuoka pointed out speaker's belief or assertion of what he says appears at the end of the sentence [14]. For example, Japanese particles and auxiliaries at the sentence-final positions show speaker's judgment on what he says expressing how probable (i.e. /kamoshirenai/ ("may")) or how obligatory (i.e. /nakyadameda/ ("must")). The final particle shows speaker's attitude expressing confirmation or emphasis (/ne/ or /yo/). Using utterances containing these postpositionals and final particles referred in [14] (Table I), we analyzed the correlation between communicative prosody and subjective impressions given by the lexicons. Especially, to measure impression about speaker's judgment given by the modality of judgment, we selected 3 axes related to speaker's judgment, "convinced", "assertive", and "advising". The analyses showed weak negative correlations between F0 rising at phrase-final mora and the magnitude of impressions expressing the speaker's judgment, convinced, assertive, and advising, obtained from particles and auxiliaries [9]. The correlations were found only in communicative speech but not in reading speech. In this study, to understand sentence-final prosody scientifically, we employed the command-response model proposed by Fujisaki [10] to represent observed F0 contour as parameters associated with linguistical factors. Employing this F0 generation model, we tried to find F0 control characteristics through its model parameters in communicative speech.

III. CHARACTERIZATION OF JAPANESE COMMUNICATIVE FO CONTOUR

In this section, we first briefly explain well-known command-response model [10] and an additional introduced accent command for quantitative analysis of sentence-final prosody variation together with speech data employed for the analysis.

A. F0 Contour Model

The command-response model is known as F0 contour generation model relating to linguistical factors [10]. This model generates F0 contour as a sum of phrase component, accent component, and base F0 parameter F_{min}, shown in (1). α, β, and γ are constants typically $\alpha = 3.0$, $\beta = 20.0$, and $\gamma = 0.9$ respectively.

$$\ln F_0(t) = \ln F_{\min} + \sum_{i=1}^{I} A_{pi} G_p(t - T_{0i}) + \sum_{j=1}^{J} A_{aj} (G_a(t - T_{1j}) - G_a(t - T_{2j})) \quad (1)$$

$$G_p(t) = \begin{cases} \alpha^2 t\, e^{-\alpha t} & \text{for } t \geq 0 \\ 0 & \text{for } t < 0 \end{cases} \quad (2)$$

$$G_a(t) = \begin{cases} \min(1-(1+\beta t)\, e^{-\beta t}, \gamma) & \text{for } t \geq 0 \\ 0 & \text{for } t < 0 \end{cases} \quad (3)$$

In these equations, F_{min} represents F0 baseline. A_{pi} and T_{0i} represents magnitude and occurred time of phrase commands. A_{aj}, T_{1j}, T_{2j} represents magnitude, onset time, and offset time of accent commands. As shown in these equations, F0 contour is generated by quite small number of parameters which can be directly associated with prosody control factors given by the input sentence. For this reason, we adopted this model for our scientific F0 analysis.

B. Applying Additional Commands for Phrase Final F0 Control

Previous study showed sentence-final communicative prosody shape varies depending on lexical impressions [9]. From the analysis, weak negative correlation was observed between final particle (i.e. sentence-final mora) F0 rising in Japanese communicative prosody and lexical impression values obtained from postpositionals. To see and quantify the variation of sentence-final F0 shape affected by lexical impression, we introduced an additional command corresponding to sentence-final mora F0 contour. Fujisaki et al. showed final particle prosody can be

described as accent commands by dialogue prosody analysis [15].

Therefore, in this study, we represented sentence-final mora command as an accent command (hereinafter called Aalast, T1last, T2last), which shows local F0 control as shown in bottom layer of Figure 1. In Japanese utterances, accent commands are usually positive [16]. On the other hand, though the commands are usually positive as well in English, negative commands were used when speaker exaggerates para-linguistic information [16]. Since final mora F0 drop found in communicative prosody [9], we allow sentence-final command A_{alast} to be negative in the following analysis.

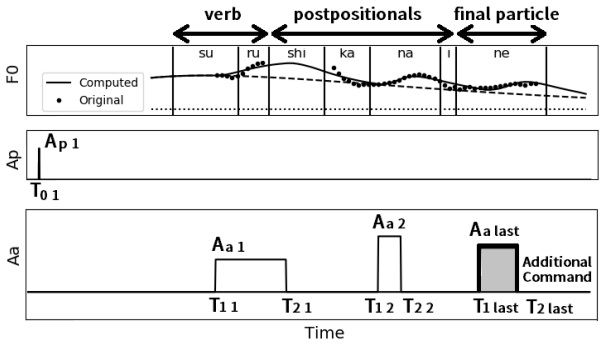


Figure 1. An additional accent command to control sentence-final prosody

TABLE I. PHRASES USED FOR ANALYSES (ENGLISH TRANSLATIONS OR EXPLANATION IN "")

Phrase form: Verb + Postpositionals + Final particle (*i.e. /suru shikanai ne/*)	
Verb (2 words)	
suru (type 0; accentless) "do"	
toru (type 1; head-accented) "take"	
Postpositional (13 words)	
working as modality of judgment; speaker's judgment about contents	
kamoshirenai "may"	*nichigainai* "must" (very likely)
mitaida "look like"	*rashii* "sound"
hazuda "should"	*bekida* "ought to"
rebaii "only have to"	*hougaii* "had better"
nakyadameda "must" (obligation)	*shikanai* "just have to"
temoii "can" (permission)	*nakutemoii* "not have to"
chadameda "must not" (prohibition)	
Final particle (2 words)	
working as modality of utterance; speaker's attitude or intention	
yo	*ne*

C. Data Sets for Experiments

Communicative speech utterances of 52 phrases with postpositionals showing different level of speaker's impression about judgment [9] were employed for the analysis. Table I shows the words employed in the phrases. These phrases consist of a verb, postpositionals working as modality of judgment, showing speaker's judgment, and a final particle working as modality of utterance, showing speaker's attitude [14]. Impression given by text of the constituting words is measured by Semantic Differential (SD) scale method [17]. Communicative speech samples were the ones uttered as the speaker talks to their friends. Their F0 contours were extracted by WaveSurfer and smoothened by simple moving average. Reading-style speech samples were also collected to compare with communicative speech. F_{min} parameters were treated as constant depending on speakers and speech styles (communicative or reading-style).

As natural communicative speech recording is quite difficult, we carefully asked all speakers to imagine real situation. Despite these considerations, as it is difficult to utter communicative speech naturally, it tends to be similar to reading speech. For that reason, we selected utterances of 10 speakers which show clear difference between communicative and reading prosody for the analysis. The difference of these prosodies was measured by F0 fluctuation range, which is residual from regression line of F0 contour.

IV. EXPERIMENTAL RESULTS

A. Controllability of Sentence-Final Communicative Prosody Based on Constituting Lexicons

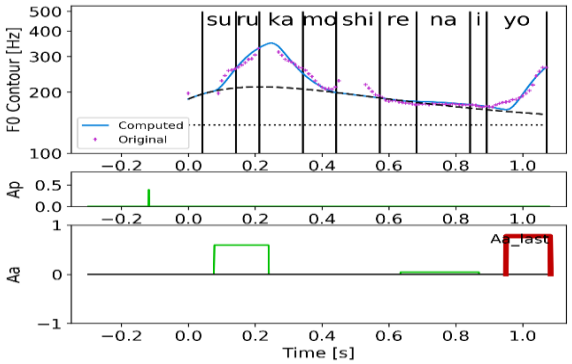
The analysis was carried out on the additional sentence-final commands in communicative prosody. The mean errors between measured F0 contours and computed ones were less than a semitone (communicative: 92 cents, reading: 80 cents). Not only positive but negative sentence-final commands were observed. As shown in the examples of Figure. 2a and 2b, sentence-final rising and falling F0 contours have been nicely approximated by the adopted sentence-final accent commands. These accent commands suggest local F0 control at the phrase-final mora occurs in communicative prosody. Also, these characteristics seem to be resulted from strong manifestation of speaker's judgment represented by "kamosirenai" (maybe) in Figure 2a and "nichigainai" (must be) in Figure 2b.

The sentence-final command magnitude A_{alast} values in communicative speech turned out to be significantly smaller than those in reading speech ($p < 0.05$). Especially, as shown in Figure 3, sentence-final

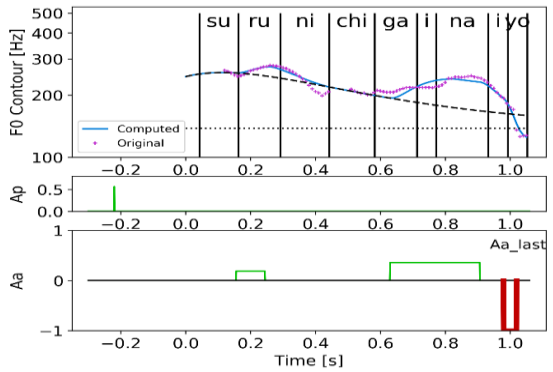
F0 drop characteristic was observed more frequently in the communicative speech (28.7%) than the reading one (17.7%).

B. Effect of Lexical Impression to Sentence-Final Communicative Prosody

In this analysis, we tried to find the possibilities to use impression obtained from lexicons for the control of communicative prosody. To focus on the lexicons constituting sentence-final parts showing judgment, we measured the magnitude of impressions about judgment ("convinced", "assertive" and "advising") obtained only from input lexicons. As we did not ask speakers to produce these speech samples with strict instructions, it is expected that individual sample and speaker may reflect factors other than impressions directly obtained from lexicons.



(a) An example of large F0 rising in speech with weak judgment (*/surukamoshirenaiyo/ (may do)*)



(b) An example of large F0 drop in speech with strong judgment (*/surunichigainaiyo/ (must do)*)

**Figure 2 .Model parameters of communicative utterances with sentence-final F0 rising (a) and lowering (b) (*bold line in accent command: sentence-*

Table II shows the correlation scores between A_{alast} and impression magnitudes about judgment obtained from constituting lexicons. As shown in the Table, weak correlations were observed in communicative prosody compared with the uncorrelated reading ones between A_{alast} and impression magnitude of constituting lexicons for the samples with sentence-final particle "yo". As Japanese final particle "yo" emphasizes speaker's judgment, these correlations imply these lexical impressions affect communicative prosody.

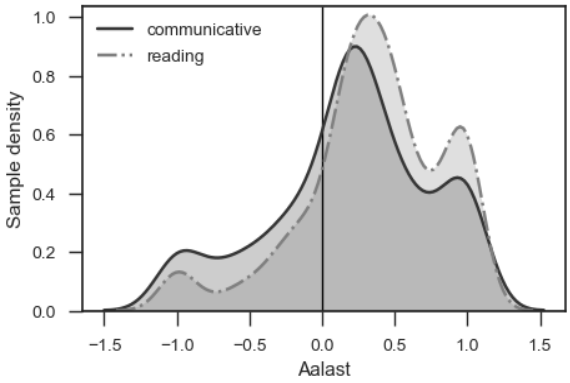


Figure 3. Distribution of Final mora command magnitude (A_{alast}) in communicative/reading speech samples

The smallness of the correlations even in communicative speech looks due to high freedom of judgment magnitude. To directly measure only the constituting lexical effect, we should strictly control the production context or apply listening-based experiments we have used in the previous study [11]. However, all correlations were minus and those in communicative prosody were larger than reading ones, which suggests the involvement of input lexicons in communicative prosody control.

TABLE II. CORRELATION COEFFICIENTS BETWEEN AALAST AND IMPRESSIONS OBTAINED FROM CONSTITUTING LEXICONS (WITH PARTICLE "YO", WHICH STRENGTHENS THE JUDGMENT IMPRESSIONS)

Correlation coefficients	Impression about judgment		
	Convinced	*assertive*	*advising*
Communicative	-0.201	-0.171	-0.241
Reading	-0.072	-0.076	-0.041

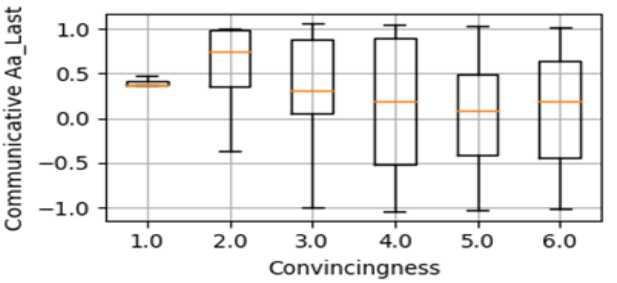


Figure .4 A_{alast} parameter within each impression value "convinced" for communicative speech samples with final particle "yo"

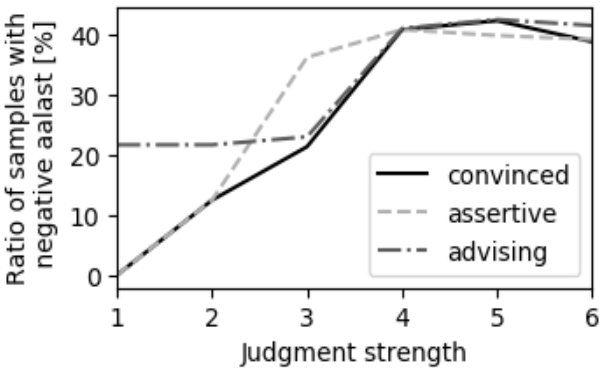


Figure 5. Ratio of samples whose A_{alast} are negative (*communicative speech samples with final particle "vo"*)

Figure. 4 shows values of A_{alast} parameters for each strength of judgment impression "convinced". Although majority of samples show positive sentence-final accent command, negative A_{alast} are also seen in utterances with strong convincingness. Figure 5 shows ratio of speech samples with negative A_{alast} command for each judgment impression strength. As shown in the figure, the negative A_{alast} command as exemplified in Figure. 2b were mostly observed in speech samples whose constituting lexicons show high scores for these three judgment impressions. In other words, the negative commands are only restricted to communicative speech samples with lexicons showing strong impression of judgment.

V. CONCLUSION

Aiming at the control of communicative prosody reflecting information obtained from input lexicons, we have analyzed F0 contours of communicative prosody using the command-response model by contrasting with reading-style prosody. For communicative speech samples consisting of single phrase with Japanese particles showing judgment magnitude ("convinced", "assertive", and "advising"), we could have observed their consistent control characteristics reflecting constituting lexical effects.

The observed control characteristics are summarized as follows.

- An additional sentence-final accent command can work systematically to express sentence-final prosody variety.
- Negative correlations were observed between final command magnitude and the all judgment impressions obtained from constituent lexicons.
- Negative control of final prosody is restricted to speech samples constituting lexicons with strong judgment.

We expect that these control characteristics enable to generate communicative speech prosody using lexical impressions. The smallness of correlations between communicative prosody and lexical impressions suggests more strict control is necessary for further analysis for computational modeling.

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Sentiment Polarity in Translation

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Abstract

Previous year, many researchers have been sentiment analysis on many focus languages. They analyzed and categorizing opinions expressed in a text. People express their opinions and feeling on social media as a daily routine. For sentiment analysis work, data plays an important role. Thus, social media become interested platform for opinion mining. On the other hand, low resource languages face less of sentiment resources (such as sentiment lexicon, corpus) than English language. It is needed to overcome language barriers and realize a sentiment platform capable of scoring in different languages when global opinion is need to decide something. In this paper, the expectations and limitations of machine translation in sentiment polarity task for Myanmar language is presented. We experiment with comments of particular news and general news that are expressed in social media news pages. Results show that sentiment transfer can be successful through human translation. This also demonstrates that translation from Myanmar to English has a significant effect on the preservation of sentiment by using translation engine. This happens primarily due to nature of Language but the results show that machine translation quality plays the important role in this work.

Keywords: *sentiment polarity, machine translation, sentiment transfer, language barriers*

I. INTRODUCTION

During these years, researchers and business industries have developed a platform for detecting opinions about the products, markets, stocks and news as expressed in social media and other media sources. They extract content from social media sites, blogs, news sites and analyze sentiment associated to each content. By doing like this, researchers can know the needs of customers, condition of specific market in particular region, public opinion on government's activities, strong and weak points of competitors and can get other benefits for industries, governments and organizations. For opinion mining work for around the world, there should not be restricted to a specific language and locale even though content appear in different languages. For successful sentiment analysis work, the system should overcome language barrier and provide sentiment analysis across languages.

At the present time in Myanmar, most people use Facebook and they read news from Facebook's news pages. They also write their opinions as comments in relative news articles using Myanmar Language. But the time of widespread information flow, we also need to listen and know opinions from other countries on our country news and events. We have already known that people from other countries use English language or their mother language to communicate the global. Therefore, at this point translation plays main role in sentiment analysis work to understand global phenomena on target work. After translating their focus language to English and then doing sentiment analysis on it that can get more opinion analysis results for particular domain from global as much as the content available for the works. Moreover, many tools developed for English such as sentiment lexicon, normalization tools or other that are not available for other languages yet. However, we should not do translation of sentiment words of one language to English without knowing how translation impact on sentiment analysis work. We should observe translation effect on sentiment work. If the machine translation system is not very accurate, there will be noise in the polarity words translated from the source language.

In this work, human translation and google machine translation are used for analysis. Human translation is very expensive and it takes time for preparation. It is not possible for many source languages. But google supports bilingual translation works. High quality machine translation from a target

language to English can eliminate the necessity to develop specific sentiment analysis resources for that language.

There are two news Facebook comments datasets for Myanmar language are used in this work. In the first one comments are extracted from 21st Century Panglong Conference news articles. The second one contains comments from general news articles. Firstly, sentiment analysis on target language is done. And then target corpus is translated to English using human translation and google machine translation system. Second, translated corpus is apply for sentiment analysis on English language. And then we investigate how sentiment polarity appear in these works. The paper is organized as follows. In the next section related work for sentiment analysis in translation is briefly described. Section 3 contains motivation and contribution of this work. Data collection and preparation is presented in section 4. Section 5 provides methodology and section 6 describe experimental results of this work. The final section contains a discussion of the obtained results, some remarks and issues that remain to be addressed and that we intend to investigate in future work.

II. Related works

As mention above, many researchers did sentiment analysis (SA) research on various languages. SA is the computational study of opinions, sentiments and emotions as they are expressed in text. A. Balahur and M.Turchi [1] expressed the impact of machine translation (MT) on sentiment analysis in French, German and Spanish. They employ three MT systems for comparison- Bing translator, Google translator and Moses. The performances of the sentiment analysis on original language and translated corpora were comparable. In the worst case, the performance difference reached 8%. The approach in [2] experiments that polarity-annotated datasets in English and Turkish for movie and products reviews domain. The authors concluded that the polarity detection task is not affected by the amount of noise introduced by MT. The publication [3] illustrate the impact of Machine Translation on sentiment analysis. The authors presented the development and evaluation of Spanish resources for the multilingual sentiment analysis tool SentiSAIL. And then they described empirically the impact of MT on sentiment analysis performance. The performance decrease in the worst case remained within negligible 5%. They drew conclusion as an outcome of the experimental setup is that substituting multilingual sentiment analysis by English sentiment analysis via MT may be an acceptable alternative. The authors of [4] present a lexicon-based sentiment analysis system in Spanish, called Sentitext, which used three feature sets- the dictionary of individual words, the dictionary of multiword expressions and the set of context rules. They concluded that multiword expressions are critical for successful sentiment analysis. Sentitext is also used in [5] to detect sentiments on Twitter messages in Spanish.

Work on multilingual sentiment analysis has mainly addressed mapping sentiment resources from English into morphologically complex languages or from target language to English language. The authors [6] translated Chinese customer reviews to English using a machine translation system. The translated reviews are classified with a rule-based system applied on English lexicons. A high accuracy is achieved by combining knowledge from Chinese and English resources. The publications [7] and [8] did sentiment analysis and normalization work on Myanmar language. In [7] researchers created domain-specific lexicon for opinion mining work. Based on experimental results that lexicon is suitable for analysis work on particular domain.

Recent research papers show that google translate has a good performance to European languages but relatively poor in Asian languages. Even thought, the effort in this paper takes a different direction as it evaluated English sentiment analysis applied to translated data from Myanmar language.

III. Motivation and Contribution

In this golden age of social media, comments and posts on social media is a part of people's daily routine. Social media allow people to express their feeling, opinion and debate for many users and many topics. In Myanmar, most people use Facebook for many purposes. Government and organizations post news on Facebook pages and every news journal and media groups also post news on their Facebook pages. Thus, many people read news from their favorite pages and write their opinion on news article as comments. Therefore, news from social media pages are attractive point for analysis of opinion of people. Most Myanmar people write comments using Myanmar language and some people use English. On the other hand, in the globalization age, we need to mine opinion of people from international news pages and should consider perception of people on Myanmar news articles from other countries. In the global news site, users use English or their native

languages. Therefore, machine translation and sentiment analysis on English language plays a major role for capturing their opinions on news articles. In this paper sentiment analysis on news domain for Myanmar language is done. Myanmar sentiment words are translated into English language using Google translate. And then we did sentiment analysis on English language again. After these steps, we have surveyed translation effect on sentiment work. Another fact for this work is that sentiment analysis tool for English language can be more available for low-resources language such as Myanmar and other target language. Therefore, I motivated to analysis how sentiment is preserved after machine translation. In this work, sentiment annotated corpus for Myanmar language for news domain is updated. There are two datasets. One dataset, 21st century Panglong Conference, has already built. Another dataset contains comments from general news articles from popular news pages in Myanmar. Contributions in this paper are as follows: (1) Data preprocessing step and building annotated general news corpus of Myanmar language. (2) English sentiment labeled corpus is created. (3) Polarity is calculated using annotated corpus and point wise mutual information method. (4) we investigate which translations that preserve sentiment best. This is a crucial step towards multilingual sentiment platform for news domain.

IV. DATA COLLECTION AND PREPROCESSING

The internet is an active place with respect to sentiment information. From a user's view, people are able to post their own content through various social media and blogs. From a researcher's view, many social media sites release their API, allowing data collection and analysis by researchers and developers.

A. Data Collection

In this work there are two datasets. Data for these two datasets are extracted from eight Myanmar popular news Facebook pages according to socialbaker.com: 7Day News Journal, Eleven Media Group, BBC Burmese, MRTV-4, Mizzima-News in Burmese, The Irrawaddy-Burmese Edition, VOA Burmese News, DVB TV News. Detail data analysis and description of 21st century Panglong Conference dataset has already mentioned in [7]. It contains 27,337 comments from news pages. Average length of the comments in this dataset is 21 words and the average number of words is 238,532 words. Another dataset contains comments on general news articles from 1st September,2019 to 31st October,2019 extracted from pages. The second dataset contains 656 comments from 98 news articles and average number of words is 13,214 words. Average words of the comments are 15 words. For classification task, the text fall into one of the following three classes: positive, negative and neutral.

B.Data Collection

Nature of social media users, they write status and comments using informal text. The dataset is examined for better understanding of the nature of collected data from Facebook. According to analysis, they use abbreviating (short form or acronym), mix usages (Myanmar and English words mixing eg. today နေသာတယ်-It's sunny today), slang words(ကွီး-ကိုကြီး-brother), multiword expressions (အေးပါပါပါပါ-အေးပါ-ok), emotion icons, syntactic mistake and using myanglish (nay kg lar? -how are you? -နေကောင်းလား). There are four types of comments in collected data :(1) Comment does not contain sentiment words. (2) Neutral comments (cannot consider in classification work. In this work PMI value is zero, sentence is noted as neutral sentence) (3) Comments suitable for sentence level but it uses many conjunction words. (4) Comments contain sentiment words. Example and explanation of each type of comment is shown in Table 1.

TABLE I. TYPE OF COMMENT IN COLLECTED DATA

Comment Type	Example	Explanation
1.	ကဗျာတွေစာတွေရေးလို့နေ- writing poems and letters	Cannot extract sentiment words from comment.
2.	တကယ်-really?	Cannot conclude polarity for it.
3.	ကြိုးစားနေပါသော်လည်း အလုပ်လုပ်နေပါသော်လည်း ပြည်သူတွေရဲ့နှလုံးသားကို မရလျှင်ကြိုးစားသမျှသဲထဲရေသွန်- Even if you try hard but work, but don't get the hearts of the people, pour it into the sand	contains two positive sentiment words -ကြိုးစား(try hard), အလုပ်လုပ် (work hard),One negative words- မရ- not get But comment gives negative meaning

4.	ငြိမ်းချမ်းခြင်းရဲ့အရသာ-the taste of peace	Positive sentiment word (ငြိမ်းချမ်းခြင်း-peace)

C. Preprocessing

Online data have flaws that potentially hinder the process of sentiment analysis. Many preprocessing steps are applied on the available dataset to optimize it for further experimentations. There are three steps in preprocessing on collected data.

1) Data cleaning process: New types of words such as emoticons (☺ and <3), hashtags("#Bieber"), URLs, photos or stickers comments are presented in social media data. Data cleaning processing is done to eliminate the incomplete, noisy and inconsistent data.
2) Syllable segmentation process: Myanmar language does not have boundary word markers. Syllable segmentation is done on cleaning dataset by using syllable segmentation tool.
3) Words Extraction process: Syllable segmented dataset is tokenized applying the n-gram method by setting the minimum and maximum grams. According to our analysis result on 500 comments, bigrams is set as minimum and 5-grams as maximum gram.

In the previous publication works [7] and [8] detail process for online data preprocessing is presented.

V. METHODOLOGY

In this work, we are interested to study the impact of machine translation on Myanmar sentiment analysis. The whole experimental stage is outlined below:

1) Build the Myanmar general news dataset MM_data and use previous domain specific dataset MM_ Panglong_data.
2) Learn a SA system on the two datasets based on pre-learned manually annotated Myanmar SA lexicon. This is denoted as the MMSA_baseline.
3) Test the MMSA_baseline system on two test dataset MM_data and MM_ Panglong_data and compute its performance.
4) Translate MM_data and MM_ Panglong _data from Myanmar to English. The English version of these datasets are obtained.
5) Learn a SA system on the two datasets based on pre-learned manually annotated English SA lexicon. This is denoted as the MSA_MT.
6) Test the MSA_MT system on two test datasets and compute its performance.
7) Compare the performance of both systems MMSA_baseline and MSA_MT and draw and investigate some conclusions.

A. Pointwise Mutual Information (PMI) based Sentiment Classification

Pointwise mutual information (PMI) is used for sentiment classification. It is a measure of association used in information theory and statistics. In computational linguistics, PMI has been used for finding collocations and associations between words. For example, counting of occurrences and cooccurrences of words in a text corpus can be used to approximate the probabilities $p(x)$ and $p\ (x,\ y)$ respectively. This method calculates the PMI between two words to obtain numeric score. The formula is as follows.

$$\text{pmi}(\text{x};\text{y}) = log\frac{p(x,y)}{p(x)p(y)} \quad (1)$$

$$\text{pmi}(\text{w1};\text{w2}) = log\frac{prob(w1\&w2)}{prob(w1) * prob(w2)} \quad (2)$$

where, $prob\ (w1\&w2)$ is the probability of word1 and word2 co-occur in the comments. Sentiment orientation score is calculate using $pmi(word1, positive\ word)$ and $pmi(word1, negative\ word)$.

$$Scorepmi = pmi\ (word1, positive\ word) - pmi(word1, negative\ word \quad (3)$$

$Scorepmi$ is calculate for all phrases. If score value is positive then comment sentence is categorized as positive, if value is negative then sentence is categorized as negative and if value is zero, it categorized as neutral.

B. Creating Sentiment Labeled Data in Myanmar and English

Manual sentiment annotations were performed on the dataset. MM_ Panglong _data dataset has been annotated by previous work. In this work, only MM_data dataset is annotated sentiment label by manually. To annotate English sentiment label, we translated Myanmar words to English by human and using google translate. MM_ Panglong _data dataset

is larger than MM_data dataset. Thus, we select 400 Myanmar sentiment words from MM_ Panglong _data for translation. And then we annotated sentiment label on these translated datasets. As mention above, there are three classes for annotating sentiment labels: positive, negative and neutral.

In the publication [7] has already created sentiment lexicon for 21st century panglong conference news articles. This lexicon is updated by adding some words from general news articles. English sentiment lexicons are freely available for research work. At the present time, we cannot apply these available lexicons. Thus, English sentiment lexicon is created by manually. This lexicon contains 101 positive words and 106 negative words. In the future, we will use freely available sentiment lexicons for English language and construct automatic sentiment lexicons by using the classification methods.

C. Impact of Translation on Sentiment Analysis

In this section, how to get translation words from Myanmar words to English words and how this translation is impact on sentiment analysis work is presented. There are two types of translation is done in this work. The first one is human translation and another one is by using google translation engine. Firstly, we need to know the quality of machine translation engine for Myanmar language. And then We can survey how English translation of Myanmar text alters or not in detection of sentiments. To evaluate translation quality for the translation engine, BLEU scores are calculated. Using human translation as the reference. By comparing machine translation to human translation, we can know how close it is to the human translation.

On the other hand, human translation work is very expensive and time-consuming process. Thus, human translation is done for randomly selected 400 comments for 452 words as a training dataset. At the same time, these selected comments are also translated by using google translation engine. Preprocessing for these sentences is done by segmenting the Myanmar sentence using Myanmar word segmenter which is described by [7]. Google translate is primarily designed to translate sentences; it can also provide one-word translations. Therefore, google translate is used to translate Myanmar words into English in comments.1-gram BLEU score for selected sentences is 0.801 and 5-grams BLEU score is 0.791.

The 1-gram is used to assess how much information is maintain after translation. Among other grams, 5-grams is the most correlated by human translation by human. At the present time, google performed a little poor quality in Asia language. However, BLEU is not sufficient for sentiment analysis. It only evaluates translation quality based on human translation. Therefore, we need to do other experiments for this work. Compare the sentiment labels assigned to the translated English text with manual sentiment annotations of the Myanmar text. The more similar the sentiment annotations are, the less is the impact of translation. Some extracted words are shown in table2 which are impact on the performance of sentiment analysis.

TABLE II. SOME EXAMPLES WHICH ARE IMPACT ON THE PERFORMANCE OF SENTIMENT ANALYSIS

comment, sentiment label	Translated by google	Translated by manually
အေးပါကွာ, *positive*	(It' s cool, *neutral)*	ok, positive
ပြောသွားတာ_က_(ကောင်းပါတယ်, positive)လက်တွေ့_ဘဲ_(လိုနေတာ, negative)	It' s good (positive) to say that It needs(negative) to be practical	good, positive need(negative) to be practical,
မလိမ့်တပတ်, *negative*	Not a week, *neural*	lie, negative
(ညာနေတာပါ, *negative*) _ကွာ	I' m right, *positive*	lie, negative
ခန့်ညားနေတာ, *positive*	It' s ugly, *negative*	Neat and tide, positive

Some Myanmar words need to normalized to improve in translation. Eg. (ဆာမ-sir (wrong translation meaning because spelling mistake in Myanmar word), if this word is normalized to the word-ဆရာမ(can get correct translation-teacher)). Out of vocabulary words in the translated text are marked as unknown words. Most of these words are slang words and abbreviating words. Sentiment class of the first comment in table2 changes positive to neutral because google translate directly by input words. But actually, it meaning is depend on content of articles. In the second comment, there are two sentiment words. Google translate gives correct translation for word by word translation for this comment. If the whole sentence is given as an input to google, translated meaning is different (missing sentiment word). (ပြောသွားတာက ကောင်းပါတယ်လက်တွေ့ဘဲလိုနေတာ-well that's good). Therefore, in this work, we

translate from Myanmar to English by segmented word by word. For the remaining three comments, translated words effect to decrease the performance of sentiment analysis. At the present time, human translation can give more accurate translated word than google from Myanmar to English language. But for the big dataset, it is very expensive, time consuming and not practical.

VI. Experimental Results

For the experimental results the two training datasets are used. The first one is 21st century Panglong Conference dataset named dataset1 which include comments from one activity, Panglong Conference article. The second one is comments from general news articles within two months which is named as dataset2. Dataset1 contains 17,337 comments and dataset2 contains 656 comments. Myanmar sentiment lexicon is constructed words extracted from 10,200 comments from both datasets. To evaluate the accuracy, we followed 3-fold and 5-fold cross validation process on these two datasets. For this experiment, we selected only 400 comments from two datasets. Because this experiment requires more manually work and human resources. To keep the time, we choose this small size of data comments.

Firstly, we investigate whether translations can maintain or not sentiment polarity from Myanmar text. For this work, sentiment lexicons for both languages are used. Firstly, sentiment analysis on Myanmar language is performed. 3-fold and 5-fold cross validation results are performed for investigation. To answer the question that human translation preserves sentiment score or not, we did sentiment analysis with human translated sentences.

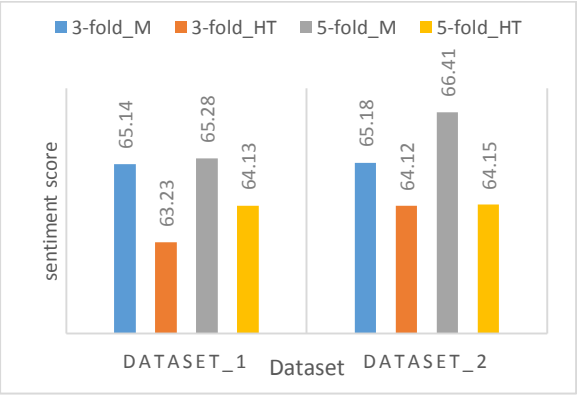

Figure 1. Sentiment Comparison: Myanmar and Human Translated English Language

If there is no significant difference between the sentiment score of Myanmar and English translation, we can conclude that the sentiment value did not change so much in translation work; if not the sentiment value is already lost in human translations. According to figure1, there is no significant difference between Myanmar and human English translation words.

According to 3-fold and 5-fold analysis results, difference between Myanmar sentiment score and human translated English sentiment score are not large. Thus, we can draw conclusion that human translation can preserve sentiment from original text. Second, we need to know the performance of machine translation engines on sentiment preservation. We compared the sentiment of the Myanmar with the sentiment of machine translation. We found that there were significant differences between this pairs. The results are shown in figure2.

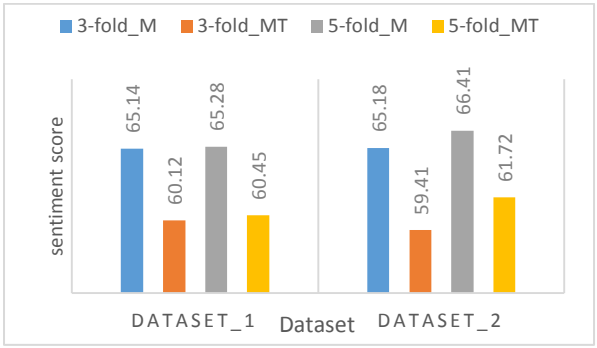

Figure 2. Sentiment Comparison: Myanmar and Machine Translated by google

In this experiment, the performance decrease in the worst case remain between 5% and 6%. We notice that both human and machine translation have lower sentiment score than original text. But we can ignore the difference between human translation and original text. One of the reasons for performance decreasing is that translations changes some words' sentiment to neutral value. Actually, these words have positive or negative sentiment values. Because translation lose content information of the articles. Other fact that complex nature of Myanmar language and most social media users use many slang words and grammar mistakes in comments. Moreover, other native languages have more than one machine translation engine such as Google, Microsoft. But Myanmar language has only one translate engine, Google. At the present time, the results indicate that translations generated by machine engine are not of the desired high quality and losing sentiment value. We can conclude that human translation is successful to preserve sentiment value. Thus, later we can use human translation as benchmark to compare other translation system.

VII. CONCLUSION AND FUTURE WORK

In this paper, we presented experimental results to study the impact of English machine translation on sentiment analysis of Myanmar comments on social media data. The results based on these datasets show that sentiment analysis on English translation by google translation engine is not good as sentiment analysis of Myanmar language at the present time. But English translation reach competitive results by using human translated English language. Thus, we observed that the quality of translation system is very important for sentiment analysis work on across languages. The important fact that this approach fails to take into account the divergence in the expression of sentiments across languages and content of articles. Moreover, according to this analysis 3-fold and 5-fold, accuracy is not completely depending on the size of training and testing data, but somewhere by increasing training data to get higher testing data accuracy. Other perspective is to investigate the use of deep learning classifiers, CNN in sentiment analysis work for better performance. In the future we will integrate other machine translation system and normalization process to enhance the performance and will evaluate the performance on data from multilingual social media platform. Moreover, we will use freely available English sentiment resources for comparative performances.

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Time Delay Neural Network for Myanmar Automatic Speech Recognition

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Abstract

Time Delay Neural Network (TDNN) contains in neural network architectures. In Automatic Speech Recognition, TDNN is strong possibility in context modeling and recognizes phonemes and acoustic features, independent of position in time. There are many techniques have been applied for improving Myanmar speech processing. TDNN based acoustic model for Myanmar ASR in this paper. Myanmar language is a low resource language and no pre-collected data is available. A larger dataset and lexicon than our previous work are applied in this experiment. The speech corpus contains three domains: Names, Web News data and Daily conversational data. The size of the corpus is 77 Hrs and 2 Mins and 11 Secs and include 233 female speakers and 97 male speakers. The performance of TDNN for Myanmar ASR is shown by comparing with Gaussian Mixture Model (GMM) as a baseline system, Deep Neural Network (DNN) and Convolutional Neural Network (CNN). Experiments evaluation is used 2 test data: TestSet1, web news and TestSet2, recorded conversational data. The experimental results show that TDNN outperforms GMM-HMM, DNN and CNN.

Keywords: *GMM-HMM, DNN, CNN, TDNN, acoustic modelling*

I. INTRODUCTION

Automatic Speech Recognition (ASR) aims to enable computers to "understand" human speech and convert it into text. ASR is the next frontier in intelligent human-machine interaction and also a precondition for perfecting machine translation and natural language understanding. It contains two models; the language model models probabilities of word sequences and the acoustic model describes distributions of acoustic features for individual phones. Typically, the two statistical models are independently trained from large volumes of text data and annotated speech data, respectively. The component connecting these two models is the pronunciation lexicon mapping words into phone sequences [1].

ASR requires an acoustic model that can effectively learn the context of adjacent input speech features to improve the recognition performance [2]. Nowadays, almost all many speech recognition systems have been used neural networks to achieve recognition performance.

Neural Networks are requires many speech data in convergence of model training, and as a result, it consumes a lot of time in model training. Time-delay neural network (TDNN) [3] is utilized to model long-term dependencies. Sequence classification is transformed to multidimensional curve classification by TDNN.

Neural network architectures have been applied to advantage for speaker adaptation. iVectors is used to extract information about speaker or acoustic environment, which have been exposed to be valuable for instantaneous and discriminative adaptation of the neural network [4]. iVectors-TDNN based acoustic model is applied in this work to improve the performance of Myanmar ASR and the results are evaluated using our own Myanmar speech corpora, UCSY-SC1 [5].

There are some recent works on Myanmar ASR using different Machine Learning approaches. H. M. S. Naing, et. al., [6] presented a large vocabulary continuous speech recognition on Myanmar language, applying three acoustic models of One Gaussian Mixture Model (GMM) and two Deep Neural Networks (DNNs) on 40 hours of speech dataset. The evaluations were done on an open test set of 100 utterances, recorded by 25 Native speakers. Word Error Rate (WER) reached up to 15.63 % in the sequence discriminative training DNN.

A. N. Mon et. al., presented CNN based Myanmar ASR, building Myanmar speech corpus, 20 hours read speech recorded by 126 females speakers and 52 males speakers, and evaluated with two datasets, opened-data, web news data and closed-data, recorded data, and achieved 24.73% and 22.95% of Word-Error-Rates[7].

A speech corpus, UCSY-SC1, is introduced in [5], and it is evaluated by comparing GMM-HMM, DNN and CNN, for improving Myanmar ASR, showing their experimental results, 15.61% and 24.43% (WER).

TDNN-based acoustic model is applied for Korean corpora [8] shows has an advantage in fast-convergence on TDNN when the size of training data is restricted, as sub-sampling excludes duplicated weights. and not as an independent document.

II. AUTOMATIC SPEECH RECOGNITION

Main Component of Automatic Speech Recognition system are Pre-processing, Feature Extractor, Acoustic Model, Language Model and Recognizer. An acoustic model's task is to compute the P (O|W), i.e. the probability of generating a speech waveform for the model [9].

A. GMM-HMM Model

GMM is a probabilistic model which is signified as a biased amount of Gaussian element densities and used to model the distribution of the acoustic characteristics of speech. Gaussian distribution is intended by mean, variance and weight. HMM is used to represent the transition probabilities between states.

The transition between phones and corresponding observable can be modeled with the Hidden Markov Model (HMM). An HMM model composes of hidden variables and observables [10]. The horizontal arrows demonstrate the transition in the phone sequence. As shown in Fig. 1 [16].

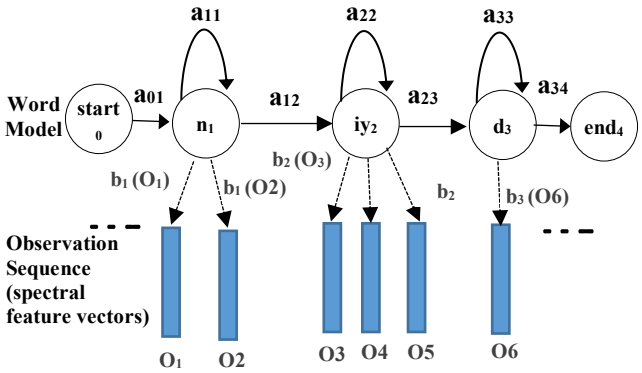

Figure 1. Typical HMM architecture

GMM-HMM system, 39 features of employed for training and testing of this system using MFCC feature extraction technique. It contains 13 static energy of first and second changing derivations order delta features and cepstral mean and variant normalization. The implementation is done on 25 and 10 ms frame length and with a frame shift of 10ms is sliced from 9 frames with combination and project down into 40 feature vectors, followed by applying linear discriminant analysis (LDA) to lower 40 dimensions. The predictable vectors are handled with maximum likelihood linear transform (MLLT) method to produce de-correlation. The feature space maximum likelihood linear regression (fMLLR) is applied in this work for speaker adaptive training (SAT). The preparation of these vectors are combined with GMM-HMM classifiers. The training is started with state of a decision tree for tri-phone model and mono-phone model.

B. Deep Neural Network (DNN)

Deep Neural Networks are many hidden layers of units between input and output layers. They are typically feed forward neural networks (FFNNs). DNNs contain many hidden layers with a large number of non-linear units and a large output layer. The large output layer is performance as an input to various number of HMM states. DNN-HMM outperformed traditional GMM-HMM due to increased modeling power [11].

For DNN-HMM based model, feature space MLLR (fMLLR) method is applied in this experiment. Mel-Filter Bank features estimated in speaker adaptive training approach and is worked on GMM based system. Four hidden layers with three hundred units per hidden layers used in this work. The process is started high dimension 1024. Same process of fMLLR transformation is done in development and testing phase which is applied in decoding module.

C. Convolutional Neural Network (CNN)

CNNs are popular models of deep learning that are widely used in ASR. They are decreasing spectral variations and modeling spectral correlations in acoustic features. Hybrid speech recognition systems including CNNs with HMM-GMM have accomplished the state-of-the-art. CNNs consists of weight sharing, convolutional filters, and pooling. Therefore, CNNs have reached an impressive performance in speech processing. CNNs are composed of many convolutional layers. Pooling is that decreases the dimensionality of a feature map [12]. The same setting of CNNs [5] is applied in this work for evaluation.

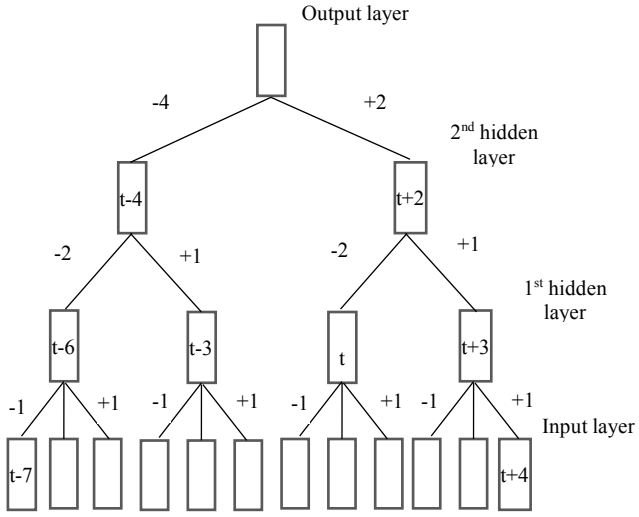

Figure 2. Typical structure of TDNN architecture

D. Time Delay Neural Network (TDNN)

Time-delay neural networks are designed to express a relation among inputs in time for speech recognition. TDNN design the initial transforms are learned within narrow temporal contexts while the later layer operate a wider temporal context [8]. Example of TDNN architecture is shown in Fig. 2 [17].

TDNN architecture are tied across time steps, which is to decrease parameters and learn shift invariant feature transforms. Another way is used to reduce parameters and advance calculation is sub-sampling, for example, the splicing configuration {-1, 1} means that splice the input at present period substract 1 and the current time step add 1 [2] [8]. Input context {-1, 1} is applied in this experiment. 100-dimensional iVector to the 43-dimensional input apply on each frame. 11 hidden layers and bottleneck-dimension is 256 and a constant learning rate is 0.001. The number of epochs 20 and mini-batch-size 128 is used in this work.

III. PRONUNCIATION LEXICON

The acoustic model uses phonemes and language model is output words, a lexicon is needed to connection the gap. The lexicon is used to translate words or sentences into a sequence of phonemes. The lexicon needs to able to map each word in the language model to a sequence of phonemes. A Grapheme-to-Phoneme (G2P) converter can automatically transcribe the words needed for the lexicon.

This experiment of lexicon was created by training a G2P conversion model using Myanmar Dictionary. The dictionary includes in total about phonetically transcribed words. This paper used a lexicon that has the word size of 44,376 that is the extension of Myanmar Language Commission (MLC) [13]. There are 110 phonemes in the training set.

IV. SPEECH DATASET

UCSY-SC1 [5] formed by combining data with web news and conversational data. This speech corpus contains three domain: Names, Web News and Daily conversational data. Names are unique 2,250 sentences and obtained from UCSY website[1], which is recorded with 10 intern students. Conversation data have been collected from UCSY NLP Lab members and Internship students with 58 speakers and have over 7,200 unique sentences. Names and Daily conversation are recorded with voice recorder Tascam DR-100MKIII[2]. Sample frequency used with 16 kHz. 832,658 words and 15,095 unique words in this corpus. Web News Data are from Myanmar News Websites such as MRTV news, Eleven News, For Info News, 7Day TV. TestSet1 and TestSet2 are Web News and conversation data. The statistics of corpus is shown in TABLE I.

TABLE I. STATISTICS OF SPEECH CORPUS

Data	Size	Speakers			Utterances
		Female	*Male*	*Total*	
Names	6Hrs 30Mins	6	4	10	22,250
Web News	25Hrs 20Mins	177	84	261	9,066
Daily Conversation	45Hrs 3Mins	42	4	46	72,003
TestSet1	31Mins 55Secs	5	3	8	193
TestSet2	32Mins 40Secs	3	2	5	887
Total	77Hrs 2Mins 11Secs	233	97	330	104,399

TABLE II. INPUT FEATURES

Acoustic model	No. of Hidden layers	Learning rate	No. of feature dimensions (mfcc)
DNN	4	0.008	40
CNN	4	0.008	40
TDNN	11	0.001	43

The details input features are used in this work for neural network experiments as shown in TABLE II.

V. EXPERIMENTAL SETUP

This experiment developed with 76 Hrs and 53 Mins training data. SRILM language modeling toolkit [14] is used to build language model. The details statistics are shown at TABLE III. The development dataset was taken from training set. Dataset contain two sets: TestSet1 and TestSet2, which are the same with UCSY-SC1 [5] corpus. TestSet1 is open test data, which is Web News. It is generated by 8 speakers. TestSet2 is also open test set with native 5 speakers recorded to the conversational data with recorders. For all the neural networks training is worked on TESLA K80 GPU. The system experiments are done using Kaldi [15] toolkit version for development.

The evaluate of speech recognition systems is measured word error rate (WER) [7], which it can then be computed as:

$$WER = \frac{100 \times (insertions + substitutions + Deletions)}{TotalWord \in CorrectTranscript} \quad (1)$$

[1] http://ucsy.edu.mm/

[2] https://tascam.com/us/product/dr-100mkiii/top

TABLE III. STATISTICS ON TRAIN AND TEST SETS

Data Setup	Size	Speakers			Utterances
		Female	*Male*	*Total*	
Training	76Hrs 53Mins	225	92	317	103,319
Developm-ent	30Mins 12Secs	6	4	10	213
TestSet1	31Mins 55Secs	5	3	8	193
TestSet2	32Mins 40Secs	3	2	5	887

TABLE IV. EXPERIMENTAL RESULT

Model	Dev (%WER)	TestSet1 (%WER)	TestSet2 (%WER)
	Speaker Independent	*Speaker Independent*	*Speaker Independent*
GMM-HMM	21.50	26.11	28.78
DNN	15.84	20.20	22.60
CNN	15.55	18.44	20.50
TDNN	**11.25**	**15.03**	**16.83**

VI. EXPERIMENTAL RESULT

Myanmar ASR performance is evaluated using TDNN acoustic modeling technique and compared with baseline GMM-HMM, DNN, CNN model as shown in TABLE IV.

TABLE IV shows word error rates of Development data (Dev), TestSet1 and TestSet2 based on training data. Four acoustic model are compared. This experiment use the same lexicon and same language model. TDNN is outperforming than 4 acoustic models in speaker independent.

VII. CONCLUSION

This paper presents the extension of speech corpus UCSY-SC1 [5] for Myanmar ASR. The large speech corpus is to support for Myanmar speech development because Myanmar is a low-resourced language. GMM-HMM, DNN, CNN and TDNN are applied to evaluate the performance of extended speech corpus. TestSet1 and TestSet2 are used to estimate the ASR accuracy. TDNN is better than base line acoustic model GMM-HMM, and DNN and CNN. Applying sub-sampling in TDNN decreases the model size that reduces the number of parameters for the hidden layers, so that the dimensions of hidden layers decreased significantly. TDNN also leads to the lowest error rates on both TestSet1 and TestSet2.

Building the speech corpora is important for low-resourced Myanmar language and expected that this corpus will be some of use for more Myanmar speech processing. End-to-End learning approach will be applied in our future work to improve the performance of Myanmar Automatic Speech Recognition.

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University Chatbot using Artificial Intelligence Markup Language

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Abstract

Chatbots are conversational systems that can do chat interactions with human automatically. It is developed to be virtual assistant, making entertainment for people, helping for answering the questions, getting driving directions, serving as human partner in smart homes etc. Most of the chatbots utilize the algorithms of artificial intelligence (AI) in order to get the required responses. In this paper, we provide the design of a University Chatbot that provides an efficient and accurate answer for any user questions about university information. This is the first University Chatbot for inquiring about school information in Myanmar Language based on Artificial Intelligence Markup Language and uses Pandorabots as the interpreter.

Keywords*—AIML, Natural Language Processsing, Pandorabots, Pattern Matching, Response Generation*

I. INTRODUCTION

Conversational agents become essential by interacting of machines with the desired users to provide natural language interfaces. So, the role of chatbots in the information technology and communication is widely in used. Many chatbots are created day by day through marketing, medical, education and banking. Chatbot is also a user assistant substance that is intended to produce a communication with human through their regular language. In educational system, it is essential for teaching, learning and searching the desired information for a specific area. The obvious factor that leads us one step closer to living in our fantastic world is that it knows our messages and can respond to us. The bot would match the input sentence from the user with that pattern existed in the knowledge base. This system is simple Myanmar chatbot using AIML but it can answer the necessary information for the users. There are many chat engines with different methods and can perform chatting. Some famous chatbots are SimSimi, Mitsuku, A.L.I.C.E, and now the machine learning chatbots like Siri, Alexa, Cortana and so on. Although modern chatbots apply the power of artificial intelligence to answer complicated questions, they still need some improvements for low resource languages.

II. LITERATURE REVIEW

B. A. Shawar and E. Atwell [2] described a system to access Arabic language information using chatbot without sophisticated natural language processing or logical inference. They showed this work properly with English and European languages and how the same chatbot will react in terms of Arabic FAQs. S. A. Prasetya, A. Erwin[3] aimed to found right AIML interpreter which can be used in Indonesian language E-Commerce website. The system is built by using Artificial Intelligence Markup Language (AIML) and Pandorabots as the interpreter for the customers tending to buy things online. S. Hussain, O. A. Sianaki, and N. Ababne[4] discussed chatbots classification, design techniques used in earlier chatbots and modern ones, and how the main categories of chatbots can handle conversation context. They presented with the emergence of new technologies intelligent systems have emerged using complex knowledge-based models. B. R. Ranoliya, N. Raghuwanshi and S. Singh[5] provided the university related FAQs for students by using the Artificial Intelligence and Latent Semantic Analysis. They implemented to solve the academic needs of the visitors for Manipal University. A. Mondal, M. Dey, D. Das, S. Nagpal, K. Garda[9] focused on the design of texual communication in educational domain. They developed and processed the accurate chatbot by using random forest algorithm. R. Sharma, M. Patel[10] presented the review of design techniques on chatbot in speech conversation systems. They discussed the performance and usability of many chatbots in our everyday lives.

III. HUMAN COMPUTER INTERACTION IN CHATBOTS

Human Computer Interaction is a communication field of study focusing on the design of computer technology and the interaction between human and computers. Conversation system between a human and a computer is either chatting by typing text or speech dialogue using the voice. Thus, interaction with natural language is a feasible option for connecting machine agents and human users. The chatbot popularity has brought the new feeds for HCI as it has changed the pattern of human interaction. Human Computer Interaction may need to consider for chatbots as the main object of design, focus on services than user interfaces, and design for interaction in networks of human and machine actors[13]. The main parts which include human computer interaction in conversation systems design are (a) the techniques used to produce keywords, (b) pattern matching techniques used inside the chatbot and (c) the type of response. So, HCI is considered taking on human-chatbot interaction design as an area of research and practice.

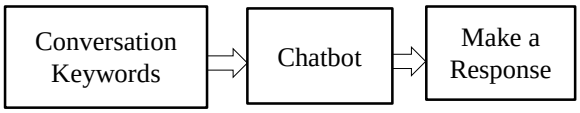


Figure 1. Human Computer Interaction

IV. ARTIFICIAL INTELLIGENCE MARKUP LANGUAGE

AIML is an XML based markup language for specifying chatbot content. It was created by the ALICE bot free software community in 1995-2000 for the people to input dialogue pattern knowledge into chatbots based on the ALICE free software technology. An AIML Interpreter is able to load and run the bot, then, it provides the bot's responses in a chat session with a user. AIML consists of data objects called AIML objects, which are made up of units called topics and categories. The topic is called an optional top-level element, it has a name and a set of categories related to that topic. Categories are the basic units of knowledge in AIML. One category is a rule for matching an input and converting to an output, and consists of a pattern, which represents the user input, and a template, which responses the answer. The AIML pattern is simple and consists of words, spaces, and the wildcard symbols _ and *.

```
<?xml version="1.0" encoding="UTF-8"?>
<aiml version="2.0">
<!—AIML code goes here -->
</aiml>
```

A. AIML Categories

There are three AIML types: (a) atomic categories, (b) default categories, and (c) recursive categories[1].

(a) Atomic categories are those with patterns that do not have wildcard symbols, _ and *.

```
<category>
<pattern>မင်္ဂလာပါ</pattern>
<template>ဟုတ်ကဲ့ မင်္ဂလာပါ ရှင့်</template>
</category>
```

(b) Default categories include wildcard symbols * or _.

```
<category>
<pattern>ကျောင်းချိန် က ဘယ်အချိန်မှာ * </pattern>
<template>ကျောင်းချိန် က မနက် ၉ နာရီမှာ စပါတယ်
</template>
</category>
```

For such situation, if the user enters ကျောင်းချိန် က ဘယ်အချိန်မှာ စတာလဲ then the AIML class will search until the wild symbol (*) and if there is a match it will accompany response.

(c) Recursive categories are the categories with templates <srai> and <sr> tags, which represent recursive artificial intelligence and symbolic reduction. Applying a combination of wild cards and srai, the stop words of the sentences can be carefully checked out from the user input. Recursive categories involve many applications: **(i) symbolic reduction** which reduces the complex grammatical forms to simpler ones; **(ii) divide and conquer** category splits an input into two or more subparts and add the responses to one; **(iii) synonyms resolution** is possible to appear different words with the same meanings depending on the consisting text; **(iv) keyword detection** is possible to find the same response when a definite keyword is found in the user input [6]. These are some examples of different AIML categories.

(i) symbolic reduction

```
<category>
<pattern>တီချယ်သီ ကို သိလား</pattern>
<template>ကွန်ပျူတာသိပ္ပံမဟာဌာန မှ ဌာနမှူး
ဖြစ်ပါတယ် </template></category>
<category>
<pattern>တီချယ်မွန်း ကို သိလား</pattern>
<template>သုတသိပ္ပံမဟာဌာန မှ ဌာနမှူး ဖြစ်ပါတယ်
</template></category>
<category>
 <pattern>သင် * ကို သိလား</pattern>
<template><srai><star/> ကို သိလား</srai>
</template></category>
```

(ii) divide and conquer

```
<category>
<pattern>ကျေးဇူးပါ ရှင့်</pattern>
<template>ဟုတ်ကဲ့ ပြန်လည် တွေ့ဆုံဖို့ မျှော်လင့်ပါတယ်
</template> </category>
<category>
<pattern> ကျေးဇူးပါ *</pattern>
<template> <srai>ကျေးဇူးပါ ရှင့် </srai>
</template></category>
```

(iii) synonyms resolution

```
<category>
<pattern>ကျောင်းလိပ်စာ သိချင်လို့ပါ</pattern>
<template>အမှတ်-၄ လမ်းမကြီး၊ ရွှေပြည်သာမြို့၊
ရန်ကုန်တိုင်း ဖြစ်ပါတယ်</template>
</category>
<category>
<pattern>ကျောင်းက ဘယ်မှာ ရှိတာလဲ</pattern>
<template><srai>ကျောင်းလိပ်စာ သိချင်လို့ပါ </srai>
 </template> </category>
```

(iv) keyword detection

```
<category>
<pattern>သင်ယူခြင်း</pattern>
<template> သင်ယူခြင်းသည် ပြောင်းလဲခြင်း ဆီသို့
ဦးတည်စေသော လုပ်ငန်းစဉ် ဖြစ်သည်</template>
</category>
<category>
<pattern> _ သင်ယူခြင်း</pattern>
<template><srai>သင်ယူခြင်း</srai></template>
</category>
<category>
<pattern>သင်ယူခြင်း *</pattern>
<template><srai>သင်ယူခြင်း</srai></template>
</category>
```

```
<category>
<pattern>_ သင်ယူခြင်း *</pattern>
<template><srai>သင်ယူခြင်း</srai></template>
```

V. PROPOSED SYSTEM

We have implemented a Myanmar interactive chatbot for university frequently asked questions. AIML is defined with general inquiries and messages which are replied by applying AIML formats. According to the Artificial Intelligence Markup Language, we have used different AIML tags to get the user required information from the bot.

TABLE I. AIML TAGS USED IN SYSTEM

No.	Tags used for AIML Categories
1.	<topic> </topic>
2.	<category></category>
3.	<pattern></pattern>
4.	<template> </template>
5.	<srai></srai>
6.	<random> </random> with
7.	<set> </set>
8.	<get> </get>
9.	<that> </that>
10.	<think> </think>
11.	<condition> </condition>

The system operation is divided in three steps. In the first step, the question is entered by the user. In the second step, the system performs word processing actions to match the user's input to a pre-defined format and do the pattern matching between user input and the Knowledge Base. Finally, the answer is presented to the user in the third step.

A. Knowledge Base

Artificial Intelligence Markup Language is a well-known XML derived language to build chatbot knowledge base. Users' frequent asked question sets are defined semantically the knowledge domain given to the chatbot. The questions are available from the university academic center related to nine topics and manually collect from the teachers, students and their parents that they want to ask about the university. We have used 970 question-answer pairs as data distribution. A well-designed knowledge base can

positively impact the effectiveness of chatbots that will improve the interaction between users.

B. Workflow of the System

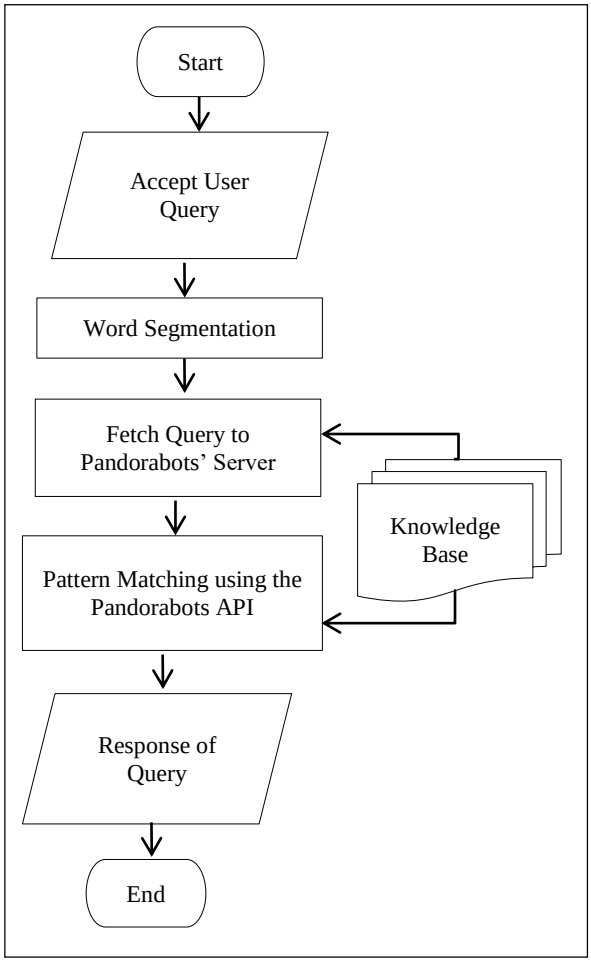


Figure 2. Flow of the System

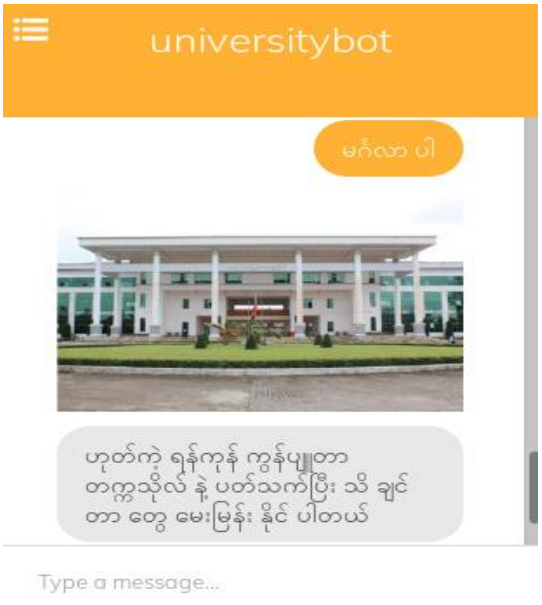
VI. Implementation of the Proposed Chatbot

Rule-based chatbot contains a faster time-to-relevance, delivering a faster impact on user interaction. This chatbot is one of rule-based chatbot and developed on AIML language for the University of Computer Studies, Yangon. We have purposed this system to have a support for university routine. All the questions files need to be uploaded to Pandorabots server. The files include the university related questions and information that the students, teachers and parents frequently asked. The number of question-answer pairs in the system that are utilized for different topics and type of categories are as given:

Table II. The Topics and Number of QA pairs used in the System

Topics	Atomic	Default	Recursive
Greeting	25	12	23
Location & Address	13	15	7
Academic	172	200	61
Brief History	18	20	15
Conference	39	35	11
Faculties & Staff	50	25	22
Library	40	20	10
Research & Lab	34	38	20
Alumni	12	25	8

The service requires internet connection to access the system. The users can interact with the chatbot in every time if there is a connection. The user needs to input the questions as Myanmar Language. The segmentation process is done by using the UCSY word segmentor (http://www.nlpresearch-ucsy.edu.mm/NLP_UCSY/wordsegmentation.html). The input from the user is normalized and processed on the Pandorabots server. The AIML files, which are separated into several categories and the chatbot's knowledge is uploaded into the server. After pattern matching, the inquired user can ask the university related questions about academic services and activities. These are some sample results of our system:



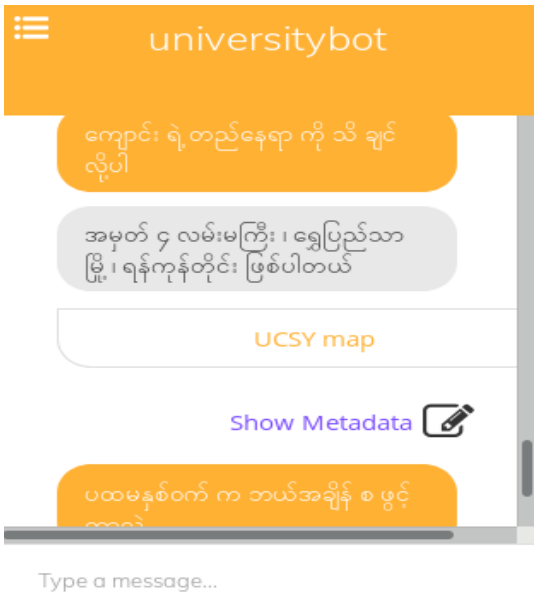


Figure 3. Sample Outputs of the System

VII. TESTING AND EVALUATION

Testing can be made to measure the quality of chatbot. The steps included to conduct the chatbot experiments are (i) getting the overviews of questions that can be asked, (ii) inquiring the user questions related to the nine topics show in Table 2, (iii) asking the questions to get the feedback about the system, (iv) the experimental analysis whether they have correct or wrong responses [12].

We have made user testing with the closed user group by students, parents and staff. The system still needs improvement due to some patterns that mismatch with the chatbot knowledge. We also have adopted the dialog efficiency matrix to evaluate the chatbot.

A. Dialogue Efficiency Metric

The proposed system is measured the efficiency with four sample dialogues in terms of atomic categories, default categories, recursive categories. Adopted learning mechanism is used to see the ability to find answers for user as shown in table III.

TABLE III. TESTING IN FOUR DIALOGUES TYPES

Matching Type	D1	D2	D3	D4
Atomic Categories	8	13	20	10
Default Categories	25	9	7	19
Recursive Categories	14	10	15	20
Total	47	32	42	49
Mismatch Pairs	3	2	3	6

In the above table, we have tested the system with four dialogues as D1, D2, D3 and D4 for three matching categories. The bot can answer most of user questions correctly. We have also found some miss-match questions and answers as in figure 4. The error may occur by the user input or may be the chatbot knowledge. When we have multiple categories with the same pattern, AIML interpreter chooses categories from bottom to top within each individual files. Unexpected bot responses can cause due to normalization error. This can be solved by the substitution file in the system. We also need to prepare more data for chatbot's knowledge.

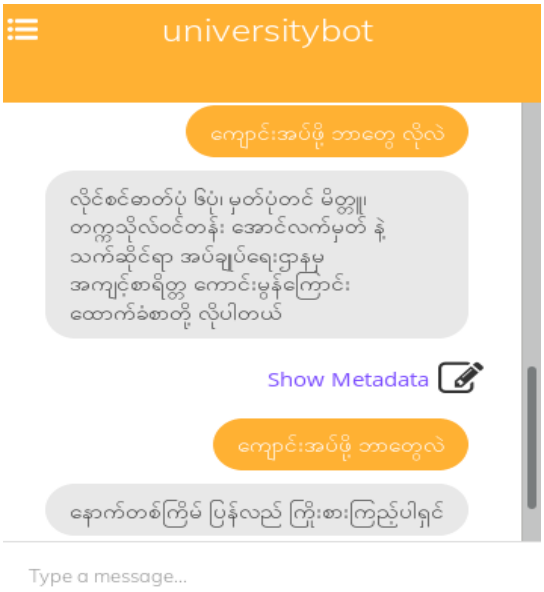


Figure 4. Mismatch Question Example

The frequency of matching type in each dialogue generated between user and chatbot was calculated in figure 5. These absolute frequencies are normalized to relative probabilities.

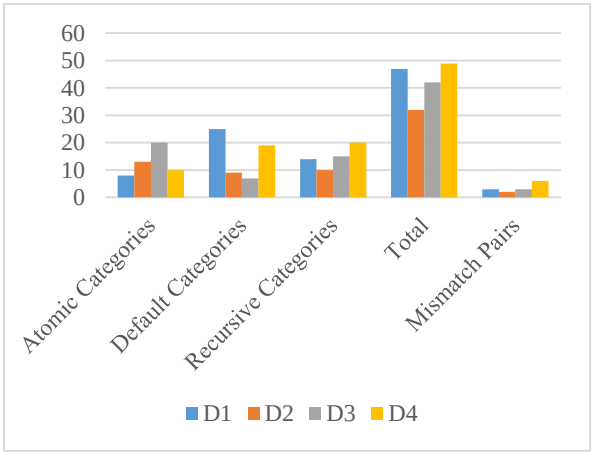


Figure 5. Dialogue Efficiency of the Tested Dialogues

VIII. CONCLUSION

Chatbots can interact with people in effective ways. There are many chatbots in English and other languages by using different algorithms and models but there is little chatbot using Myanmar language. This is one of the University Chatbots using Myanmar Language to fulfill the information gaps between the university and its related users. Now, we have implemented a chatbot for the University of Computer Studies, Yangon. This is simple chatbot using Artificial Intelligence Markup Language and implemented on the Pandorabots server. The user can ask the useful questions about the university related the academic sectors through chatbot. The bot will help people to save time and get the information every time. We still need some improvements for the bot and we will develop this by using machine learning techniques in coming jobs.

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Security and Safety Management

A Detection and Prevention Technique on SQL Injection Attacks

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Abstract

With the web advancements are rapidly developing, the greater part of individuals makes their transactions on web, for example, searching through data, banking, shopping, managing, overseeing and controlling dam and business exchanges, etc. Web applications have gotten fit to numerous individuals' day by day lives activities. Dangers pertinent to web applications have expanded to huge development. Presently a day, the more the quantity of vulnerabilities will be diminished, the more the quantity of threats become to increment. Structured Query Language Injection Attack (SQLIA) is one of the incredible dangers of web applications threats. Lack of input validation vulnerabilities where cause to SQL injection attack on web. SQLIA is a malicious activity that takes negated SQL statement to misuse data-driven applications. This vulnerability admits an attacker to comply crafted input to disclosure with the application's interaction with back-end databases. Therefore, the attacker can gain access to the database by inserting, modifying or deleting critical information without legitimate approval. The paper presents an approach which detects a query token with reserved words-based lexicon to detect SQLIA. The approach consists of two highlights: the first one creates lexicon and the second step tokenizes the input query statement and each string token was detected to predefined words lexicon to prevent SQLIA. In this paper, detection and prevention technologies of SQL injection attacks are experimented and the result are satisfactory.

Keywords—*SQL Injection Attack, Web applications, Malicious activity, Vulnerabilities, Input validation*

I. INTRODUCTION

Web application is one of the most mainstream communication streams with the rapid development of web advances. Information is a significant job in data frameworks. Many associations run their transactions on database appended web applications to get information from clients. Web application is a significant wellspring of data for any organization to get business process basic information and broadly utilized in different applications. With the ubiquity of web applications, there are numerous security issues in the web world and furthermore increment in web application vulnerabilities. SQL injection is software vulnerabilities in web applications which is brought about by absence of information approval. The information approval weakness is where user input is utilized in the product without affirming its legitimacy. SQL represents Structured Query Language, which is the standard programming language for creating social databases. It an order and control language utilized in the making, altering, erasing, and recovering the information and structures that involve the social database framework. SQL injection is one of the most serious dangers to the security of backend database from driven applications.

SQL injection is an assault method with negated SQL articulations used to abuse how site pages speak with back-end databases. It can take a shot at defenseless website pages that adventure a backend database like MySQL, Oracle and MSSQL. Attackers can give directions (made SQL explanations) to a database utilizing input fields on a site. Figure 1 shows the procedure of SQLIA. These directions control a database server behind a web application to get self-assertive information from the application, meddle with its rationale, or execute directions on the database server itself. Along these lines, the impacts of SQLIA are side step verification, extricating information, loss of privacy and respectability.

SQL injection is strikingly like a XSS. The essential contrast being that a XSS attack is executed at the web front end, though the SQL assault is executed at the server. The issue in the two cases is that client input was never approved appropriately.

There is assortment of methods are accessible to identify SQLIA. The most favored are Web Framework, Static Analysis, Dynamic Analysis, consolidated Static and Dynamic Analysis and Machine Learning Technique. Web Framework gives separating way to deal with channel exceptional characters however different assaults are not recognized. Static Analysis checks the information parameter type, yet it neglects to identify assaults with right info type. Dynamic Analysis method is equipped for

checking vulnerabilities of web application however can't distinguish a wide range of SQLIA. Joined Static and Dynamic Analysis incorporates the advantage of both, yet this technique is unpredictable so as to continue. AI strategy can recognize a wide range of assaults however results in number of bogus positives and negatives.

SQLIA can be presented with the following segment of vulnerable Java code:

```
String uname= request.getParameter("uname");
String pword= request.getParameter("password");
String sql_query= "SELECT name FROM member
WHERE username=' "+uname+" ' AND password= '
"+pword+" ' ";
Statement stmt=connection.createStatement( );
ResultSet rset=stmt. executeQuery(sql_query);
```

In this code, string variable sql_query is used for keeping the cunning SQL query statement that is being executed in the database.

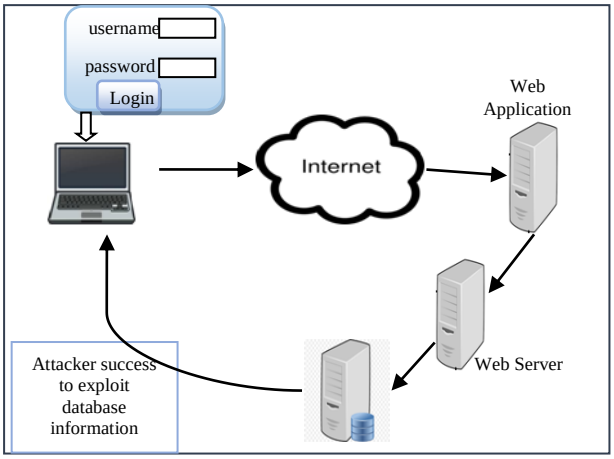

Figure 1. How SQL Injection Attack Works

II. RELATED WORKS

N. Lambert et al. [1][5] proposed a model that uses a tokenization technique to detect SQL injection attacks, so query containing Alias, Instances and Set operations can also be blocked at the entry point. It checks whether the generated query based on user's input its intended result, and compare the results by applying tokenization technique on an original query and input query. If the results are same, there is no injection, otherwise it is present. I. Balasundaram et al. [2] proposed a technique for SQLIA using ASCII based string matching. This technique is used to check the user input field with static and dynamic analysis to detect and prevent SQLIA.

M. Kumar et al. [4] and William G.J. Halfond et al, G.Yiğt, M. Arnavutoglu [14], discussed the detection and prevention techniques of SQL injection attacks and analyze existing techniques against such attacks. B.J.S. Kumar & P.P. Anaswara [15] experimented on detection and prevention of SQL injection attack. D. Kilaru [12] observed how SQL injection occurs and how to update a web app with SQL injection vulnerability. R. M. Nadeem et al. [13] proposed a system which is based on dynamic analyzer model. This model received the user request and analyzed to check that request is for pages without vulnerabilities (P') and with vulnerabilities (P), with help of knowledge base. J.O.Atoum and A.J.Qaralleh [16] described static and runtime SQL queries analysis called hybrid techniques which is to create a defense strategy that can detect and prevent various types of SQLIA.

III. TYPES OF SQLIA

In web-based applications, most of work is to access data from databases. If the user input data is not properly performed or validated, users can access information they were not supposed to get access to. The following techniques are types of SQLIA.

A. Tautologies

In tautology based attack, the general goal of attacker is to input crafted SQL tokens that cause the conditional query statement to be evaluated always true. An attacker undertakes an input field with vulnerability that is utilized in the query's WHERE predicated and transform the condition into a tautology which is always true. This predicate logic is assessed as the database examines each tuple in the table. If the predicate logic at WHERE clause is evaluated as a tautology, the database match and returns all tuples in the table rather than associating only one tuple, as it would normally do in the sense of injections. This type of attack proceeds to bypass authentication and extract data [4, 7, and 14].

Smith ' OR 'a'='a	SELECT name FROM member WHERE username='Smith' AND password= ' ' OR 'a'= 'a'
' OR 'a'='a ' OR 'a'='a	SELECT name FROM member WHERE username =' ' OR 'a'='a' AND password=' ' OR 'a'='a'
'OR 'a'='a' - -	SELECT name FROM member WHERE username =' ' OR 'a'='a' - - ' AND password=' '

This query statement is always true because it have been added by the tautology statement ('a'='a'). Double dash "--" instructs the SQL parser that the rest of statement is a comment and should not be executed

B. Malformed Queries

In this approach, when the attacker abuses an illegitimate or deficient SQL token, the rejected error message is come back from the database including helpful debugging information. The error message causes assailants to precisely distinguish which parameters are vulnerable in an application and the total outline of the underlying database. This situation was misused by assailant's crafted SQL tokens or garbage input that causes syntax mistake, type jumble or logical error into the database. To recognize injectable parameters, syntax errors can be utilized. Type errors can be applied to conclude the information kinds of specific attribute or to remove information. Logical errors can be out the table names and attribute names that cause the mistake or error [9].

C. Union Query

In this technique, attackers injected invalid statement is joined with the valid query by the utilizing UNION keyword. Attackers misuse a query statement of the structure "UNION <injected query>" as far as possible of the end of legitimate statement. It makes the application return information from the results of original query and furthermore information from another table. Then, the statement written the double dash "- -" will be comment out [11].

```
SELECT name FROM member WHERE username="UNION SELECT password FROM member WHERE username='admin' -- AND password="
```

In this query, the original query returns null set whereas the exploited query statement returns data from the same table.

D. Piggy-backed Queries

In the piggy-backed query-based attack, an attacker attempts to add extra queries into the first inquiry string. It abuses database by the query delimiter, for example, ";" to add additional query to the first query. In the event that the attack is fruitful, the database gets and executes a query statement that contains numerous particular inquiries. The first query is the original legitimate query, which is to execute the database whereas the second query, malicious query is to misuse the database [4].

```
SELECT name FROM member WHERE username='Smith' AND password="; DROP table users - -
```

The two queries were separated by the delimiter, ";", and both were executed. The second query makes the database fails to client table information. Different sorts of queries can be executed with this technique, for example, addition of new clients into the database or execution of stored procedures. It is worth nothing that numerous databases don't require a special character to isolate distinct queries, so basically examining for an exceptional or special character isn't an effective way to prevent this type of attack.

E. Inference

In inference-based technique, attackers make queries that cause an application or database to act contrastingly based on the consequences of the query. There are two well-known assault strategies that depend on inference: blind injection and timing attacks.

In blind injection, developers omit detail information of error messages. These messages assist attackers to exploit to the databases. In this case, Attackers are trying to exploit the database with the vulnerable query statement that has a boolean result. Then they analyze differences based on the applications responses.

In timing attacks, attackers collect data from database by detecting timing delays in the database's responses. It depend on the database pausing for a specified time limit, then returning the information that is indicating successful query executing [8].

```
SELECT name FROM member WHERE username= IF(((SELECT UPPER(MID(password,1,1)) FROM member WHERE username='admin') ='A'), SLEEP(5),1)
```

F. Stored Procedure

In this approach, stored procedures are victims for attackers to exploit database. Stored procedures are codes that are stored in the database and execute directly by the database engine. To activate SQLIA, attackers can create injected text to exploit this stored procedure as

```
SELECT name FROM member WHERE username='; SHUTDOWN; - - password="
```

IV. PREVENTION TECHNIQUES

A. Prepared Statement

One of the best ways to prevent from SQL injection is to use prepared statement instead of statement. The problem of SQL injection is that user's input is used as part of the SQL statement. By using prepared statement, the SQL statement uses a parameter to insert a value into the database. Instead of inserting the values directly into the statement, thus prevent the backend database from running invalidated SQL queries that are unsafe and harmful to the database.

B. Using Stored Procedures

Stored Procedures adds an additional security layer to the database other than utilizing Prepared Statements. It performs the getting away from required so that the application takes input as data to be worked on instead of SQL code to be executed. The distinction between prepared statement and stored procedure is that the SQL code for a stored procedure is composed and stored in the database server, and then called from the web application.

If user access to the database is just at any point allowed by means of stored procedure strategies, permitted access for user to legitimately get to data doesn't not need to be explicitly granted on any database table. Along these lines, the database is still safe.

C. Validating User Input

In validating, user supplied input should be used after confirming its validity. So, input validation is first to ensure the user supplied input value is of the accepted type, length, format, etc. Only the input which passed the validation process prevent data from information sources, database.

D. Limitting Privileges

Limiting privileges is the concept of restricting resources from user's accesses. When you do not need to access the important part of the database, don't connect to your database using an administrator account because the attackers might have access to the entire source of system. Therefore, to identify the authenticated user there should be used an account with limited privileges to limit the extent of harms in the occurrence of SQL Injection.

E. Encrypting Data

Unencrypted data stored in database can be stolen if missing authorization or invalidated input allows users to read the data. If attackers try to gain access to database and its table, the encrypted data value will prevent attacker to read sensitive data and any further changes to databases would have no effect.

V. PROPOSED APPROACH

There are many different techniques for detecting the SQL injection attacks. The proposed technique is based on sanitizing the query statement. This approach consists of two steps: the first one is tokenizing the user inputted query. The tokenization process is made by detecting a white space, double dashes (--), sharp sign (#) and all strings before each symbol are tokens. In second step, after the query is tokenizing, each string token was detected with the contents of predefined lexicon. The contents of lexicon are most of reserved words (commands) and some logical operators. The contents of lexicon are collected most of injected commands or words in SQL injection attacks. The following table, Table I describes some words of lexicon's contents. There are 20 words in it. When the input query statement enters, whether to detect injection or not. During execution, the inputted data are matched with the corresponding lexicon's contents to check for validity. If there are matched to any other words, there is an attempt to SQL injection attack. If no, there is not an injection.

TABLE I. SAMPLE CONTENS OF LEXICON

No	Injected command
1	alter
2	concat
3	drop
4	delete
5	execute
6	sleep
7	shutdown
8	union
9	or
10	if

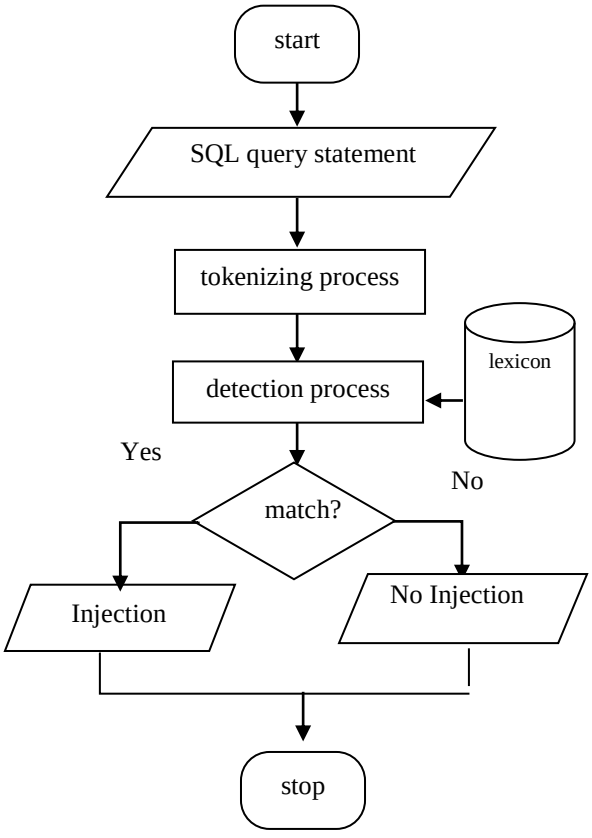

Figure 1.Flow of Proposed Approach

TABLE II. ALGORITHM FOR PROPSED SYSTEM

Algorithm: Detection

```
begin
Input:  SQL query statement
file= read contents in lexicon
N=Tokenize the query statement
flag=false
while (!eof(file))
{
        flag= false;
         for (int i=0; i< N; i++)
            if (token[i]== word in lexicon)
                flag=true;
}
if (flag) print "Injection detected";
else print "No injection";
end
```

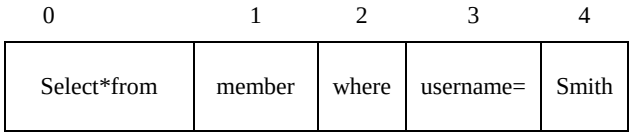

Figure 2. Tokening result without injection

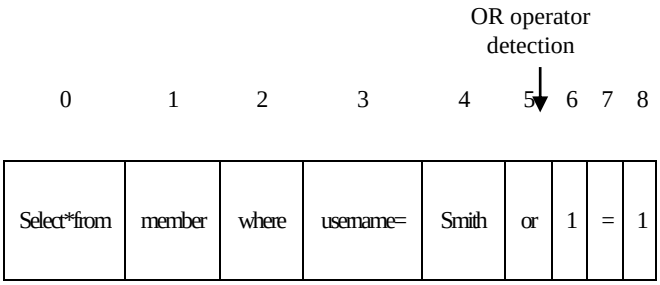

Figure 3.Tokening result with injection

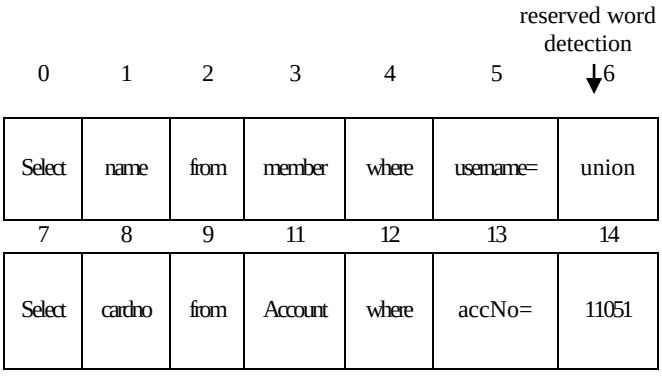

Figure 4. Tokening result with injection

The main concept of this approach is to detection to SQLIA by searching each token in predefined word of lexicon which causes to WHERE condition is always true.

VI. EXPERIMENT

In this section, the system performed some SQL injection queries on vulnerable query statement. The system tested 10 SQL query statements. These are 3 normal statements and 7 injection statements. The result outcomes describe in Table IV. The false positive and true negative scores are 0 and 7 respectively and accuracy is very good. The following table, TABLE V presents the outcomes of analysis which have evaluated. Therefore, the proposed system is done for successful prevention from various malicious query for injections.

```
String uname= "alice";
String pass= "alice123";
String query= "SELECT *  FROM member WHERE username=' "+uname+' " AND password= ' "+pass+" ' ";

uname= "alice";
pass= " ' or '1=1 ";
String query_bad= "SELECT *  FROM member WHERE username=' "+uname+' " AND password= ' "+pass+ " ' ";
```

Display Query Statement

```
Normal:
select * from member where username= 'alice' and password= 'alice123'

Injection:
select * from member where username= 'alice' and password= " or '1=1'
```

The normal query statement is no problem, as the proposed system will get data from this *member* table that satisfy the predicate. However, the approach detects the injection attack with crafted SQL statement.

```
String uname= "'; DELETE FROM member WHERE 1 or username = '";
String query= "SELECT *  FROM member WHERE username=' "+uname+" ';
```

Display Query Statement

```
Injection: SELECT *  FROM member WHERE username='';  DELETE FROM member WHERE 1 or username = ' '
```

When the system run this query, the injected delete statement would completely empty the *member* table. This system also detects the statement before to execute.

TABLE III. SQL QUERY STATEMENT

input	SQL statement
smith 123	SELECT * FROM member WHERE username ='smith' AND password ='123'
' or '1=1 ' or '1=1	SELECT * FROM member WHERE username =' ' or '1=1' AND password =' ' or '1=1'
smith ' or 'a'='a	Select * from member where username ='smith' and password =' ' or 'a '=' a'
' or ''=' ' or ''='	SELECT * FROM member WHERE username =' ' or ' '=' ' AND password =' ' or ' '=' '
smith ' or ''='	SELECT * FROM member WHERE username ='smith' AND password =' ' or ' '=' '
' or '1=1'-- 123	SELECT * FROM member WHERE username ='' or '1=1' --' AND password ='123'
'"; DELETE FROM member WHERE 1 or username = '";	SELECT * FROM member WHERE username=' '; DELETE FROM member WHERE 1 or username = ' '
"; SHUTDOWN; - -	SELECT name FROM member WHERE username="; SHUTDOWN; - - password="
Smith "; DROP table users - -	SELECT name FROM member WHERE username='Smith' AND password="; DROP table users - -
john john123	SELECT * FROM member WHERE username = 'john' AND password ='john123'
blake blake123	SELECT * FROM member WHERE username = 'blake' AND password ='blake123'

TABLE IV. OUTCOMES OF QUERY STATEMENTS

		Prediction		Total
		normal	*injection*	
Actual	*normal*	3	0	3
	injection	0	7	7
		3	7	10

$$Accuracy = \frac{TP+TN}{TP+TN+FP+FN} \quad (1)$$

TABLE V. EXPERIMENT OUTCOMES

SQLIA Techniques	Proposed approach's outcomes
Tautologies	Successful prevention
Malformed queries	Successful prevention
Union queries	Successful prevention
Piggy-back queries	Successful prevention
Inference	Successful prevention
Stored procedure	Successful prevention

VII. CONCULSION

SQLIA is one of the dominant threats to web application. Web applications need to protect their database from varying number of threats in order to provide confidentiality and integrity. In SQLIA, intruders allow attacking with a crafted query statement through a web input form into the system and then theft identity, access to sensitive information, and tamper with existing data, which can cause many disastrous effects. This paper is presented on the techniques of SQLIA and prevention approaches. The proposed approach is used for the detection and prevention of SQL injection and also suitable the outcomes.

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A Hybrid Solution for Confidential Data Transfer Using PKI, Modified AES Algorithm and Image as a Secret Key

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Abstract

Nowadays the provision of online services by government or business organizations has become a standard and necessary operation. Transferring data including the confidential or sensitive information via Internet or insecure network and exchange of them is also increased day by day. As a result, confidential information leakage and cyber threats are also heightened. Confidential information trading became one of the most profitable businesses. Encrypting the data is a solution to secure the data from being exposed. In this paper, we would like to propose a solution for the secure transfer of data using symmetric encryption, asymmetric encryption technologies and Key Generation Server as a mixed hybrid solution. A Symmetric encryption, modified AES algorithm, is used to encrypt data. Digital certificate is used both for data encryption and digital signing to assure data integrity. Key generation server is used to generate the second secret key from the publicly recognized information of a person and this key is used as a second secret key in the modified AES. The proposed hybrid solution can be utilized in any applications that require high confidentiality, integrity of data and non-repudiation.

Keywords—*hybrid encryption, encryption, PKI, modified AES*

I. INTRODUCTION

With the rising use by businesses and government agencies of the Internet as a major communication tool, digital data sharing is increasing rapidly, leading to security breaches, cyber bullying, and cyber-attacks. Many applications or systems do not have sufficient security implementation and confidential or sensitive data are accessed by intruders or hackers. Secret sharing methods protect the sensitive data from attackers.

Today, e-Government gained more and more attention and can provide a non-stop government information service through the introduction of online applications G2C, G2B and G2G. However, there are many e-government issues that need to be examined carefully, such as security issues, e-government service requirements, e-government model, e-government strategy and policy, and e-government. [1].

Cryptography, encryption techniques and security products are introduced to provide confidentially of data. Public-key cryptography is based on the asymmetric key model. By using the public-key cryptography and digital certificates, the communicating parties can authenticate each other without sharing secret information in advance. With public-key cryptography, each person gets a pair of keys, a public key and a private key. The public key is published by each user, while the private key is kept secret [2]. With the support of X.509 digital certificates in servers as well as browsers and many other communication equipment, digital certificates are used for authentication. It becomes a security solution for Internet or intranet applications to provide data integrity and confidentiality of the data.

For symmetric key cryptography, both for encryption and decryption, only one secret key is required. Asymmetrical encryption makes use of a couple of keys, a non-public (private) key and a public key. In public key cryptography, different but related pair of secret key is used both for encryption and decryption. Both asymmetric and symmetric encryption have advantages and disadvantages. And the asymmetric encryption is slower than symmetric encryption.

A new hybrid encryption approach is used in this work which combines symmetric cryptography, asymmetric cryptography together for safe exchange of data. Asymmetric encryption is used to hide the secret key of the symmetric encryption algorithm. Symmetric encryption algorithm we used in our work is a modified version of Advanced Encryption Standard and it is named AES-R. AES-R algorithm uses two secret keys and the additional secret key will be generated by the Key Generation Server. Traditional AES key will be generated from an image. Digital signature technology is used to facilitate the data integrity and non-

repudiation. Our proposed work combines all the best of the different technologies for use in real time applications.

II. DIGITAL SIGNATURE TECHNOLOGYy

Digital signature becomes a standard for cryptographic protocol suites, and it is widely used for the proof of authenticity and non-repudiation of transactions, data integrity, and communications. Financial transaction, online banking, e-Government applications, software distributions, auditing and Contract Management Software are now using digital signatures to detect forgery or tampering of the information.

Data signature technology can ensure that: information cannot be revealed by other parties except the senders and receivers; information will not be tampered during transmission; the recipient is able to confirm the identity of the sender; sender information for their own cannot be denied [3].

A. RSA Public Key Encryption

RSA public key encryption was introduced by Ronald Rivest, Adi Shamir, and Len Adleman in 1977. It is a block cipher encryption. Unlike the symmetric key encryption, key used for encryption and decryption is different, however they are closely related.

If we denote the public key as PK and other secret key (private key) as SK, Sender as X and Receiver as Y, the RSA algorithm can be stated as below.

I. Encryption and Decryption

If the Sender X sends the plain text P to the Receiver Y, first Sender X encrypts the message P with PK of Y. The Receiver Y decrypts the cipher with his private key SK to get the plain text P, namely.

Decrypt YSK (Encrypt YPK (P)) = P

If the Y decrypts the cipher text with his public key, it cannot be restored into original plain text P.

Decrypt YPK (Encrypt YPK (P)) $\neq$ P

The security for RSA public key cryptography depends on how difficult it is to calculate the private key SK from the public key PK. This calculation is equivalent to a lump sum for the decomposition into two prime number multiplication factor. So far, no matter with what kind of hardware and software, decomposing a lump sum to two qualitative factors is extremely difficult. So the RSA public key encryption algorithm is still very safe [3]. As the hardware technology is improving and the unceasing research and improvement made to the RSA algorithm, RSA algorithm has been widely used by various applications to date.

III. MODIFIED ADVANCED ENCRYPTION STANDARD (AES-R)

To get better performance in encryption time, the proposed work will use a standard symmetric encryption algorithm. As the AES is the standard of now and it is sturdy sufficient to face the attacks, our proposed security solution used modified version of the AES algorithm, called AES-R. Detail design and implementation of AES-R can be seen in the research paper [4]. The encryption process of AES-R algorithm can be seen in Fig. 1.

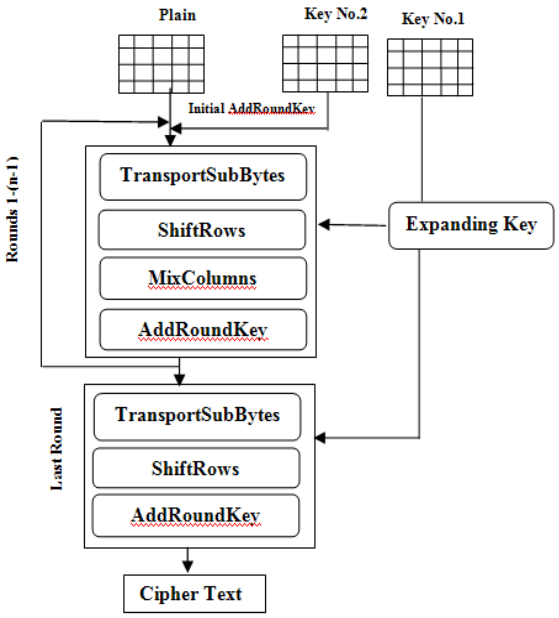

Figure 1. Encryption of AES-R

The decryption process of AES-R can be seen in Fig. 2.

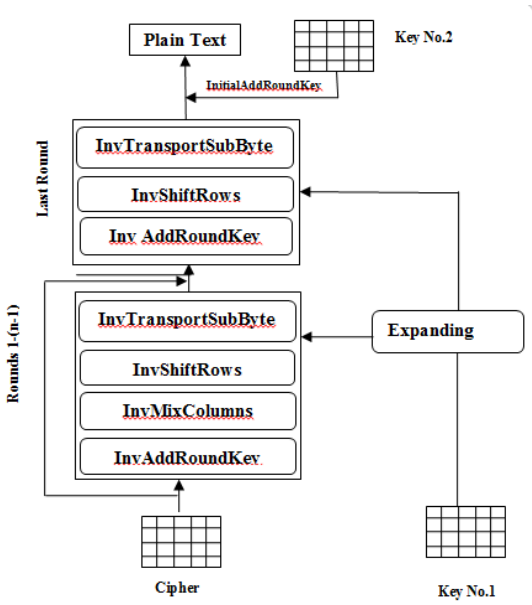

Figure 2. Decryption of AES-R

The reason AES is chosen for modification is that it can work with different key sizes and block sizes. According to the AES statement, it can accept only a block size of 128 bits and a choice of three keys; 128, 192, 256 bits [5].

IV. DESIGN OF PROPOSED WORK

Our proposed work is a combined solution that brings symmetric encryption, Public Key Infrastructure (PKI) and key generation server together.

Because symmetric encryption is faster than asymmetric encryption and since it can be implemented with hardware or software, our proposed solution has used the symmetric encryption to reduce the time for encryption. Among the symmetric encryption algorithms, AES has been a standard until now and because of its good performance, our proposed work has modified AES and proposed AES-R algorithm.

Today, the legislation of many countries recognizes an electronic document as a legal document, i.e. the same legal status as the paper documents. PKI has been established in many countries to make online identification available for online services. Certification Authorities (CAs) issue digital certificates to subscribers and the subscribers use their digital certificates for identification, authorization and digital signature signing. Currently, digital signature is the only answer used for non-repudiation of the document signer and data integrity. For that reason, our proposed work has been using digital signature technology to verify the validity of documents, and the Signer cannot refuse the transaction.

As described in Fig. 1 and Fig. 2 of AES-R algorithm, AES-R algorithm requires two secret keys. Key-1 will be used for key expanding and to generate the round keys. An image file will be used as secret key (i.e. Key-1) after it is converted into a SHA256 hash value. The flowchart of how the image will transform into a secret key(SHA256 hash) is shown in the Fig. 3.

Key-2 is an additional secret key for AES-R. To produce Key-2 of our solution, a Key Generation Server will be used. Key Generation Server will accept the user's publicly identifiable data such as public key of digital certificate, e-ID number, Passport number or e-mail as input and then it generates a secret key. Both the sender and the receiver require Key Generation Server to generate the secret key before they encrypt or decrypt the data.

A. Encryption

For the encryption, the process will perform as shown in the Fig. 4 where P is plain text, H(P) is hash of plain text, SKS is Secret Key generated from Key Server, KEY-1 is secret key of AES-R algorithm generated from the input image, C(P) is cipher of plain text P, C(KEY-1) is cipher of the KEY-1 and Rpk is public key of Receiver.

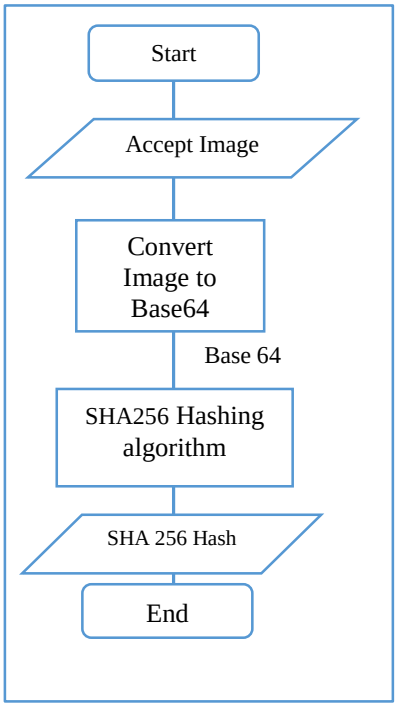

Figure 3. SHA25 Hash Generation from an Image File

To begin with, digital signature algorithm uses plain text P to generate H(P), where H(P) is the digital signature of the plain text. For confidentiality of the data, AES-R is used to encrypt the plain text by using KEY-1 and second secret key(SKS). For the confidentiality of the KEY-1, it is again encrypted with the public key of the Receiver so as to be decrypted with private key of the Receiver.

Finally, when all the encryption processes are finished, (H(P), C(P), C(KEY-1)) will be sent to the Receiver. Fig. 4 shows the encryption process done in the proposed solution.

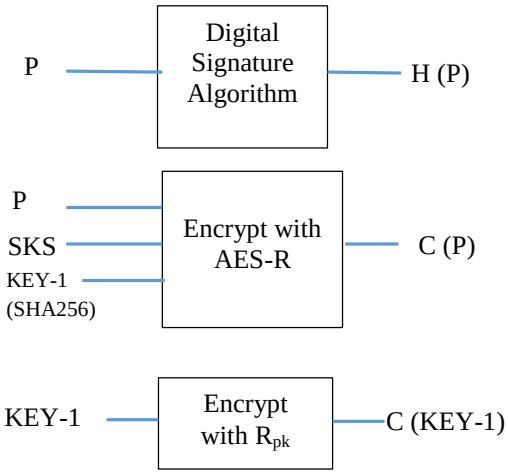

Figure 4. Encryption Process

B. Decryption

When the receiver has received the message (H(P), C(P), C(KEY-1)), the decryption process will be performed as shown in the Fig. 5. Here, Rsk is private key of Receiver and H(P)' is the new hash value generated by P after decryption by AES-R.

In the decryption process, C(KEY-1) is decrypted first to get the KEY-1. Then, KEY-1 together with secret key generated by the Key Generation Server (SKS) and C(P) are used and decrypted to get the plain text P. To verify the integrity of the data, Hash of P is generated again and we get H(P)'. Then two hash values, H(P)' and H(P), are compared. If they are equal or identical, signature is valid and data is not altered during transfer. Otherwise, the message shall be discarded.

The identical of two hash values also guarantees that data is really sent by the Sender. Fig. 5 shows the decryption process done in proposed solution.

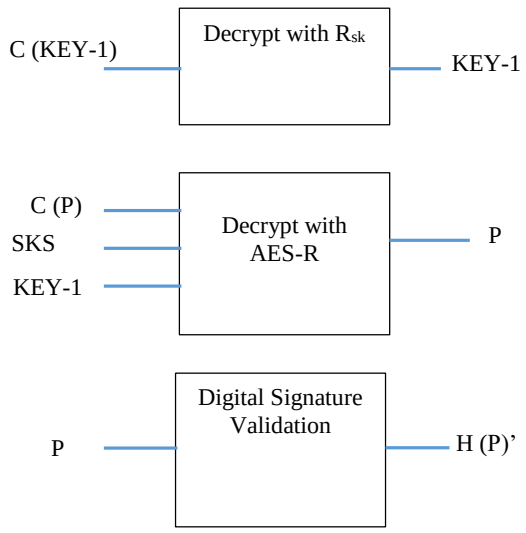

Figure 5. Decryption Process

C. Advantages of Proposed Solution

The advantages of our proposed solution are described briefly in below paragraphs.

1. Data Confidentiality

Confidentiality is the assurance of data privacy: No one can read the data except for the intended specific entity (or entities). If data or documents are transmitted over unprotected or insecure networks, confidentiality or privacy is a basic or desired requirement [6].

The proposed solution used symmetric encryption to achieve faster performance and asymmetric encryption to securely send the secret keys to the Receiver. Our proposed solution can provide confidentiality even if the information is intercepted during transfer. It cannot be restored to its original without having two secret keys and private keys of the Receiver.

2. Data Integrity

Data integrity is the assurance of no alteration whether the data is either in transit or in the storage. A digital signature provides both data origin authentication (evidence about who originated the data) and data integrity (evidence that the data has not been altered in any way) [6].

A digital signature is depending on the message because it can be computed only by the Sender's message and requires private information. As our solution use digital Signature that facilitates authentication of messages, it guarantees that no one can forge the Sender's signature. Besides that, the Sender cannot deny a message he has sent.

Our proposed work digitally signs the data so that it can be verified later with the use of digital signature technology. In addition to that, solution can facilitate identity authentication and non-repudiation.

3. Security

As we use two secret keys and a private and public key pair to encrypt and decrypt the data, it will take longer time for hackers to find the right keys than the symmetric or asymmetric solution alone. Finding the right keys to decrypt the data will be difficult and not possible with having private key of the Receiver.

V. CONCLUSION

Today, sensitive or confidential data are transferred via Internet or insecure network for business purposes or personal affairs. By using insecure network or applications, electronic documents or information may be disclosed, counterfeited, tampered or repudiated when transfer. Confidentiality, integrity and non-repudiation become a mandatory requirement for organizations and government entities.

By using the proposed solution, secret key agreement between the Sender and Receiver is not necessary. Any public information of a person can be used as one of the secret keys. The proposed work can be implemented to secure the sensitive data, for example trade confidential data, credit card information, Government's SECRET and TOP SECRET level documents, which are transferred over insecure network. The proposed solution can be used when data is shared between two people who have conflicting interests. The solution can be used to guard against fraud and it is secure than the analog signature of today.

Implementation of this proposed work will be presented as future work and its performance and impacts

will be analyzed by the view of time taken for encryption/decryption and security analysis.

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Comparative Analysis of Android Mobile Forensics Tools

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Abstract

This paper performs a comparative analysis of Android mobile forensics tools which are used for acquisition and analyzing of Android mobile devices. The major challenges of Android forensics investigation are manufacturing of Android devices with various operating system versions and there is no single tool which can be used for all sorts of Android devices. Aiming to overcome these challenges and increase more accuracy and integrity in Android forensic investigation, we made comparative analysis on both open source tools and one commercial tool. Logical and physical acquisition methods were utilized to acquire data from Android devices. Android Debug Bridge backup, Linux Data Duplicator utility tool, Magnet Acquire and Belkasoft Acquisition tools were used for acquisition. Two popular analyzing tools such as Autopsy and Belkasoft Evidence Center were utilized to analyze acquired data. The results show that using multiple tools can get more accuracy and integrity of artifacts which is forensically sound.

Keywords: *android forensics, logical acquisition, physical acquisition, forensics investigation*

I. INTRODUCTION

Digital forensics is an exciting, fast-paced field that can have a powerful impact on a variety of situations including internal corporate investigations, civil litigation, criminal investigations, intelligence gathering, and matters involving national security. While the interesting part of Android forensics involves the acquisition and analysis of data from devices, it is important to have a broad understanding of both the platform and the tools that will be used throughout the investigation. A thorough understanding will assist a forensic examiner or security engineer through the successful investigation and analysis of an Android device [1].

An investigator needs to observe about forensics tools in order to select suitable tool based on each scenario. In some cases, investigators use only certain and important data while in other cases full extraction of the physical memory and/or the embedded file system of the mobile is desirable for the potential recovery of deleted data and a full forensic examination. Therefore, the development of guidelines and processes for the extraction and collecting of data from Android mobiles is especially important, and researchers must periodically review and improve those processes according to Android technology development [2].

It is important for an examiner to understand how a forensic tool acquires and analyzes data to ensure nothing is missed and that the data is being decoded correctly. While manual extraction and analysis is useful, a forensic examiner may need the help of tools to accomplish the tasks involved in mobile device forensics. Forensic tools not only save time, but also make the process a lot easier [3]. We need to perform comparative analysis of important tools that are widely used during forensic acquisition and the analysis of Android devices.

As the use of mobile device continues to increase, it is important to efficiently acquire as much information as possible from those devices. In this work, we analyzed Android mobile forensics tools which are used for acquisition and analyzing. We focused on both open source tools and one commercial tool. The rest of this paper is organized as follows. In section 2 we reviewed some papers related with our work. Section 3 presents Android forensic methods. In section 4 we describe our experiment in detail. Comparison of forensic tools are presented in Section 5. We discuss and conclude our work in Section 6. In section 7, we plan our future works.

II. RELATED WORK

In [4], a comparative study of the Android forensic field in terms of Android forensic process for acquiring and analyzing an Android disk image was presented. The challenges of Android forensics, including the complexity of the Android applications, different procedures and tools for obtaining data, difficulties with hardware set up, using expensive commercial tools for acquiring logical data that fail to retrieve physical data acquisition were described. To solve these challenges and achieve high accuracy and integrity in Android forensic processes, a new open source technique was investigated. Manual, Logical and physical acquisition techniques were used to acquire data from an Android mobile device. Following the manual acquisition, logical acquisition was conducted using the AFLogical application in the ViaExtract tool installed on a Santoku Linux Virtual Machine. The image file is then created using the AccessData FTK imager tool for physical acquisition. Four tools were utilized to analyze recovered data: one using ViaExtract on a Santoku Linux Virtual Machine, two using the AccessData FTK Imager, and one using file carving in Autopsy on a Kali Linux Virtual Machine. The results of the analysis demonstrated that the technique can retrieve Contacts, Photos, Videos, Call Logs, and SMSs. Also, the EaseUS Data Recovery Wizard Free tool was used for the recovery of files from the LOST.DIR on external memory.

The paper [5] gives an overview of forensic software tools for Personal Digital Assistants (PDA). A set of generic scenarios was devised to simulate evidentiary situations and applied to a set of target devices to gauge how selected tools react under various situations. The paper summarized those results, giving a snapshot of the capabilities and limitations of present day tools, and also provided background information on PDA hardware and software.

In [6] the author highlighted various techniques available in the market in terms of logical acquisition, physical acquisition and analysis. They deals with the survey of various Android forensics techniques and tools. Forensics methods were discussed with respect to logical and physical acquisition process. They discussed about various tools in both the categories by studying the functionalities existing in the tools and drawbacks. Major tools are capable to provide required results for cybercrime investigation and the evidence and analysis results are acceptable in the court of law. Using these tools the tests are repeatable until unless the evidences are not tampered.

III. ANDROID FORENSIC METHODS

In forensic process, there are five phases such as identification, preservation, acquisition, analyzing and reporting. Identification is determining which device will be processed. The main purpose of the preservation is to maintain the data integrity of the device. We focus on acquisition and analyzing phases in the following sections. After analyzing phases, reporting is imperative.

A. Forensics Acquisition

Forensic acquisition is imaging or extracting data from digital devices. There are three types of acquisition methods: manual, logical and physical. The amount and sort of data that can be acquired may be different based on the type of acquisition method being utilized. Forensic acquisition tools we used in our experiment is shown in Table 1.

TABLE I. FORENSIC ACQUISITION TOOLS

Tools	Logical	Physical
Android Debug Bridge (ADB) Backup	√	
Disk Duplicator (DD)		√
Magnet Acquire	√	√
Belkasoft Acquisition	√	√

Logical acquisition is extracting allocated (non-deleted) data and it accesses the Android file system. Logical acquisition relies on Content Providers to acquire forensically sound data with effective manner. This technique can get only a fraction part of the whole Android file system.

For data acquisition, we need to use Universal Serial Bus (USB) cable to connect the mobile device to forensic workstation. After connecting, the workstation sends command to the device. These commands are interpreted by the device processor. Finally requested data is received from the device's non-volatile memory and sent back to the forensic workstation. This technique writes some data to the mobile device and may change the integrity of the evidence. With logical acquisition tools, deleted data is never accessible.

Physical acquisition is not concerned to the file system. The main advantage of this technique is it can acquire significant amounts of deleted data. When a user delete a file, it is not permanently removed by the Android system. File system only marks data as deleted, and does not actually erase the storage medium as long as there is no need more storage

space in the system. As physical forensic methods directly access the storage medium, both the allocated and the unallocated data can be obtained. Physical acquisition is generally difficult and takes long times. Wrong procedure in some steps could lead the device broken.

B. Forensic Analysis

In analyzing step, we need to extract data from acquired image file and analyze using various tools. There is no single tool which can be used for all sorts of Android device and scenario. We need to use multiple tools in order not to lost valuable information. Analysis tools we used in our experiment are listed in Table 2.

TABLE II. FORENSIC ANALYSIS TOOLS

Tools	Open Source	Commercial
Autopsy	√	
Belkasoft Evidence Center		√

IV. EXPERIMENT

The specifications of tested Android mobile devices which we used in our experiment are shown in Table 3. Samsung device has been used since 2015 and Oppo device has been used since 2018. These device are owned by one of the authors.

Table III. Specification of Tested Android Devices

Brand	Samsung	Oppo
Device Name/Model number	Galaxy Note 4/SM-N910H	Oppo A83/CPH1729
Android Version	6.0.1	7.1.1
Baseband version	N910HXXU1DPD2	M_V3_10
Kernel Version	3.10.9-7284779	4.4.22-G2019111
Build Number	MMB29K.N910HXXS2DQH5	
Micro SD Card	16 GB	16 GB

Before the acquisition is started, we isolated the Android devices from networks such as Wi-Fi, Data and Cellular to prevent changing data on the devices. And then we prepared forensics workstation installing forensics tools on a laptop computer with these specifications— Dell Intel (R) Core i7 (2.70 GHz) CPU, 8.0 GB RAM. In our experiment, after identifying the tested devices and configuring forensic workstation, we performed rooting, acquisition and analyzing using various forensic tools.

A. Rooting

On un-rooted devices, data from /data/data directory could not be accessed. Therefore, the tested Android phones were first rooted utilizing Odin3 (version 3.10.6) to upload the root-kit (CF-Auto-Root).

Installing a root-kit enables the user to gain privilege access the Android Operating System, permitting examiners to bypass a few restrictions that the manufacturers put on the device. A rooted Android phone enables the user to access protected directories on the system that hold user data (e.g., /data/data directory) and the entirety of the files in these directories. These data files can hold a lot of that may support an ongoing investigation.

B. Acquisition

We need to maintain the integrity of data by write blocking and calculating cryptographic hash value on the data. Therefore, write blocking and, Message Digest 5 (MD5) and Secure Hash Algorithm (SHA1) were used to calculate hash values and check integrity of the data.

ADB Backup. In order to acquire logical data, we used three tools such as ADB Backup, Belkasoft Acquisition and Magnetic Acquire tools. For physical acquisition, Linux DD, Belkasoft Evidence Center and Magnetic Acquire tools were used. Belkasoft Evidence Center and Magnetic Acquire can be used for both logical acquisition and physical acquisition.

To get the logical data of the device, we used the ‘adb backup’ command and also unpacked into ‘.tar’ as follow.

```
adb backup -f e:/backup.ab –shared –all
java –jar abe.jar unpack backup.ab backup.tar
```

DD tool. We used the following commands to get physical image of the tested phones. Firstly, we need to know which partition holds the data. So we used ‘mount’ command in first command window to take a look at the location of our desire data partition.

```
adb –d shell
su
mount
```

From output of ‘mount’, we knew that data is located in partition ‘mmcblk0p21’. In second command window, we did TCP port forwarding in order to transfer extracted data image to the forensics work station.

```
adb forward tcp:8888 tcp:8888
```

In first command window again, we used ‘dd’ command to get image of data partition.

```
dd if=/dev/block/ mmcblk0p21 | busybox nc -l -p 8888
```

In second command window, we used netcat.exe to transfer acquired image file to the forensics work station. Our image files were named as dd_data.dd and O_dd_data.dd, respectively.

C:\netcat\nc64 127.0.0.1 8888 > **dd.dd**

Following is alternative to transfer image file to the work station instead of TCP traffic.

dd if=/dev/block/mmcblk0p21 of=/sdcard/dd.img bs=512 conv=notrunc, noerror,sync
adb pull /sdcard/**dd.img**

We calculated SHA1 hash values and checked integrity of acquired image using these hash values.

Magnet Acquire. By using several different methods of extraction, Magnet Acquire can retrieve as much data as possible, given the enhanced security on Android. Magnet Acquire can also capture images from common storage drives. In order to obtain logical and physical images of device, we also used Magnet Acquire. It calculated MD5 and SHA1 hash values. Finally we checked integrity of acquired image using these hash values. Acquisition information of Magnet is listed in Table 4 and 5.

TABLE IV. ACQUISITION INFORMATION WITH MAGNET (SAMSUNG)

	Logical	Physical
File name	Magnet.tar	.raw (MMCBLK0) .raw (MMCBLK1)
Size	15.0 GB	29.1 GB (MMCBLK0) 14.8 GB (MMCBLK0)
Time taken	01:00:0	3:46:07

TABLE V. ACQUISITION INFORMATION OF MAGNET (OPPO)

	Logical	Physical
File name	O_Magnet.ab	.raw (MMCBLK0) .raw (MMCBLK1)
Size	5.0 GB	9.7 GB (MMCBLK0) 4.9 GB (MMCBLK0)
Time taken	00:20:0	1:15:03

Belkasoft. This tool can be used for ADB backup, Agent backup, DD backup, Odin RAM image and MTP backup. ADB backup is a method of acquiring data from an Android device that utilizes pre-installed ADB-services. Agent backup is a method for data acquisition from Android devices by collecting user data with a custom Agent-application. DD backup is a method of acquiring data from an Android device that creates a complete physical copy of its permanent. Odin RAM-imaging acquisition method is based on Odin commands utilization. We used ADB backup for logical and DD backup for physical in our experiment. Acquisition information of Belkasoft is listed in Table 6 and 7.

TABLE VI. ACQUISITION INFORMATION WITH BELKASOFT (SAMSUNG)

	Logical	Physical
File name	Blks.ab	Blks.dd
Size	1.41 GB	25.1 GB
Time taken (hh:mm:ss)	00:11:00	02:02:02

Table VII. Acquisition Information with Belkasoft (oppo)

	Logical	Physical
File name	O_Blks.ab	O_Blks.dd
Size	1.6 GB	8.3 GB
Time taken (hh:mm:ss)	00:12:00	00:40:00

C. Analysis

For analysis of acquired data from previous acquisition methods, we used Autopsy and Belkasoft Evidence Center tools. We selected 'Magnet.tar' obtained from Magnet Acquire tool for logical data analyzing because the size of this file is maximum comparing to other files. It may contain more information. We also selected 'Blks.ab' as a logical file to be analyzed. As a physical image we choose 'dd.dd' getting form DD utility.

Autopsy. Autopsy is a free and open source analysis tool. Autopsy can analyze most common Android file systems. Ingest Modules are tools built into Autopsy that can be run when the case is started, or at any point afterward. There are the 17 ingest modules in Autopsy version 4.13.0. We used 15 modules in the experiment. Even though the case is still being loaded and Ingest Modules being run, we can begin analyzing the case. In our experiment, we found nothing artifacts on 'Blks.ab'. Artifacts we have got from 'Magnet.tar' and 'dd.dd' by analyzing Autopsy tool are shown in Table 8 and 9, respectively.

TABLE VIII. ARTIFACTS OF MAGNET.TAR WITH AUTOPSY

Artifacts of Magnet.tar	Amount (Samsung)	Amount (Oppo)
Accounts	871	1161
Archives	178	59
Audio	1152	384
Contacts	615	820
Databases	653	217
Documents	1274	424
Encryption Suspected	4	5
Executable	62	20
EXIF Metadata	479	159
Extension Mismatch Detected	2115	705
Images	7872	2624
Install Applications	48	64

Keyword Hits	9782	3260
Videos	8	10

TABLE IX. ARTIFACTS OF PHYSICAL IMAGE WITH AUTOPSY

Artifacts of dd.dd	Amount (Samsung)	Amount (Oppp)
Accounts	40	53
Archives	907	302
Audio	2331	777
Call Logs	1000	333
Contacts	1023	1364
Databases	669	223
Deleted Files	37964	12654
Documents	1171	390
Download Source	79	105
Encryption Suspected	4	5
Executable	65	21
EXIF Metadata	1	1
Extension Mismatch Detected	2115	705
Images	16053	10702
Install Applications	111	148
Messages	283	377
OS Information	1	1
Videos	21	28
Web Bookmarks	10	13
Web Cookies	3213	1071
Web Downloads	152	304
Web Form Autofill	754	1006
Web History	291	388

Belkasoft. Belkasoft Evidence Center is flagship digital forensic suite. The product makes it easy for an investigator to perform all steps of modern digital investigation such as: Data acquisition from various devices and clouds, artifact extraction and recovery, analysis of extracted data, reporting, and sharing evidence. Artifacts we have got from analyzing with Belkasoft are shown in Table 10, 11 and 12, respectively.

TABLE X. ARTIFACTS OF BLKS.AB WITH BELKASOFT

Artifacts of Blks.ab	Amount (Samsung)	Amount (Oppo)
Audio	3	9
Cache	52	69
Calendar	32	42
Contacts	10	13
Cookies	1908	636
Documents	715	953
Downloads	584	778
Encrypted files	4	5
Favorites	20	6
Form values	764	254
Geo location data	4	8
Installed applications	269	358
Most visited sites	2	2
Passwords	70	23
Pictures	2515	3353
URLs	846	282

TABLE XI. ARTIFACTS OF MAGNET.TAR WITH BELKASOFT

Artifacts of Magnet.tar	Amount (Samsung)	Amount (Oppo)
Audio	4	12
Cache	204	68
Calendar	32	42
Calls	4289	1429
Chats	25642	8547
Cloud Services	232	77
Contacts	8281	11041
Cookies	3781	1260
Documents	1688	562
Downloads	581	193
Encrypted files	63	21
Favorites	24	32
File transfers	2750	916
Form values	767	255
Geo location data	54	72
Herrevad	208	69
Installed applications	272	362
Instant messengers	28720	14360
Mailboxes	967	1289
Network connections	208	277
Other files	248	82
Passwords	70	23
Pictures	22142	7380
Sessions	1	1
SMS	4821	2410
Thumbnails	18	6
URLs	780	260
Videos	35	11
Voice mail	28	56
Wi-Fi connections	47	62
Wpa_supplicant.config	47	62

TABLE XII. ARTIFACTS OF DD.DD WITH BELKASOFT

Artifacts of dd.dd	Amount (Samsung)	Amount (Oppo)
Audio	4	12
Cache	218	72
Calendar	33	42
Calls	3875	1291
Chats	25055	8351
Cloud Services	232	77
Contacts	8292	11041
Cookies	3739	1246
Documents	6250	2083
Downloads	583	193
Encrypted files	243	21
Favorites	24	32
File transfers	2687	895
Form values	767	255
Geo location data	53	72
Herrevad	208	69
Installed applications	253	337
Instant messengers	27996	13998
Mailboxes	1006	1289
Network connections	208	277
Other files	250	82
Passwords	70	23
Pictures	22332	7444
Sessions	1	1
SMS	4902	2410
Thumbnails	30	6
URLs	984	260
Videos	10	11
Voice mail	28	56
Wi-Fi connections	46	62
Wpa_supplicant.config	46	62

V. COMPARISON OF FORENSICS TOOLS

According to experiment results, we performed comparative analysis on both acquisition and analyzing tools.

A. Acquisition Tools

As ADB Backup is command line tool, forensic examiners need to familiar with commands. Android Software Development Kit (SDK) is needed to be downloaded and located in forensic workstation because ADB Backup tool is included in SDK. Acquisition time is exactly 3 hours in Samsung and 1 hour in Oppo. It took longer time than other tools.

As Magnet and Belkasoft are GUI tools, they are user friendly and easy to use. Magnet can be used per request to their team. Magnet took exactly one hour in Samsung and 20 minutes in Oppo for logical acquisition.

Belkasoft is commercial tool. We used trial version requesting to their team. For Samsung, acquisition time is just 10 minutes. Acquired size of data is only 1.41 GB. The size of data is same with one which we tried second times to be sure the size of data. Comparison of logical acquisition tools is listed in Table 13 and 14, respectively.

TABLE XIII. COMPARISON OF LOGICAL ACQUISITION TOOLS (SAMSUNG)

	ADB Backup	Magnet	Belkasoft
Type	.ab./.tar	.tar	.ab
Size	11.6 GB	15.0 GB	1.41 GB
Time (hh:mm:ss)	03:00:00	01:00:00	00:10:00
GUI	No	Yes	Yes
Cost	Free	Request	Trial

TABLE XIV. COMPARISON OF LOGICAL ACQUISITION TOOLS (OPPO)

	ADB Backup	Magnet	Belkasoft
Type	.ab./.tar	.tar	.ab
Size	3.8 GB	5.0 GB	1.6 GB
Time (hh:mm:ss)	01:00:00	00:20:00	00:12:00
GUI	No	Yes	Yes
Cost	Free	Request	Trial

DD is Linux utility tool and can be used free of charge. Examiners need to understand commands to use it. It took longer time than other tools. We can choose which partition we want to be acquired.

With Magnet tool, there is no option to choose data partition. All data of device is imaged. It divided the image file into two files: MMCBLK0.raw and MMCBLK1.raw. MMCBLK0.raw contains data from user data partition. Therefore we focused on this file. Size of the files are much larger than other tools because it imaged all data of the whole device.

Belkasoft image size is the same with DD image but time taken is only half of it. We can also choose desire partition such as user data partition. Comparison of physical acquisition tools is listed in Table 15 and 16, respectively.

TABLE XV. COMPARISON OF PHYSICAL ACQUISITION TOOLS (SAMSUNG)

	DD	Magnet	Belkasoft
Type	.dd/.img	.raw (2 files)	.dd
Size	25.1 GB	29.1 GB 14.8 GB	25.1 GB
Time (hh:mm:ss)	05:00:00	03:46:07	02:02:02
GUI	No	Yes	Yes
Cost	Free	Request	Trial

TABLE XVI. COMPARISON OF PHYSICAL ACQUISITION TOOLS (OPPO)

	DD	Magnet	Belkasoft
Type	.dd/.img	.raw (2 files)	.dd
Size	8.3 GB	9.7 GB 4.9 GB	8.3 GB
Time (hh:mm:ss)	01:40:00	01:15:03	00:40:00
GUI	No	Yes	Yes
Cost	Free	Request	Trial

B. Analyzing Tools

In analysis of logical image, 14 categories of artifacts was found by Autopsy. Belkasoft found 31 categories. In the findings of Belkasoft, there are voice mail and Instant Messages such as Viper, Facebook and Hangout. Although Belkasoft found encrypted passwords, Autopsy could not do it. Comparison of analysis tools for logical data is listed in Table 17 and 18, respectively.

TABLE XVII. COMPARISON FOR LOGICAL DATA (SAMSUNG)

	Autopsy	Belkasoft
Categories	14	31
Artifacts	25113	107004
Report	Yes	Yes
Time (hh:mm:ss)	00:39:00	00:40:00
GUI	Yes	Yes
Cost	Open source	Trial

TABLE XVIII. COMPARISON FOR LOGICAL DATA (OPPO)

	Autopsy	Belkasoft
Categories	14	31
Artifacts	9912	51237
Report	Yes	Yes
Time (hh:mm:ss)	00:15:00	00:20:00
GUI	Yes	Yes
Cost	Open source	Trial

In analysis of physical image, 23 categories of artifacts were found by Autopsy. Belkasoft found 31 categories. Although Belkasoft found encrypted passwords, Autopsy could not do it. Comparison of analysis tools for physical image is listed in Table 19 and 20, respectively.

TABLE XIX. COMPARISON FOR PHYSICAL IMAGE (SAMSUNG)

	Autopsy	Belkasoft
Categories	23	31
Artifacts	68258	110425
Report	Yes	Yes
Time (hh:mm:ss)	01:49:00	00:45:00
GUI	Yes	Yes
Cost	Open source	Trial

TABLE XX. COMPARISON FOR PHYSICAL IMAGE (OPPO)

	Autopsy	Belkasoft
Categories	23	31
Artifacts	30971	52070
Report	Yes	Yes
Time (hh:mm:ss)	00:49:00	00:30:00
GUI	Yes	Yes
Cost	Open source	Trial

VI. DISSCUSSION AND CONCLUSION

We used various free and commercial mobile forensics tools focusing on Android devices. Because of there are large number of models and manufacturer specific mobile devices, tools do not provide and have for same procedure for digital investigation process. Each tool has own procedure to acquire and analyze the data in forensically sound manner.

In acquisition process, we used ADB Backup, DD and Manget Acquire tools which are open source tools and, Belkasoft which is commercial tool. ADB Backup and DD is totally free for all users but Manget Acquire is free for only members of forensics community. Therefore, we need to request by giving our information to use Manget Acquire. For testing purpose, Belkasoft can be used for one month requesting their team. Because of ADB Backup and DD are command utilities, they are not user friendly compare to Magnet Acquire and Belkasoft which are GUI tools. ADB Backup can be used for only logical acquisition but Magnet Acquire and Belkasoft can be used for both logical and physical acquisition. ADB Backup and DD took more time than Magnet Acquire and Belkasoft for acquisition on same device. In analysis process, we used Autopsy open source tool and Belkasoft Evidence Center commercial tool. Autopsy was built with features available in commercial tools. However, Autopsy cannot extract encrypted passwords like Belkasoft can. Compare to open source tools, we can see that commercial tools can save time and, get more data and more accurate results.

According to the results of Analyzing tools, we can conclude that there are different artifacts we found based on utilized tools. Amount of data are also different. In here, we'd like to recommend Magnet Acquire for logical acquisition because it is open source and obtained much more data than Belkasoft. For physical acquisition, DD tool is more preferable because it is open source tool and can acquire data exactly like a Belkasoft commercial tool. As an analyzing tool, Belkasoft is suitable. In trial version, even though its generated reports contain only random 50% of extracted data, we have found comparable artifacts with other tools. However, there is no single tool which can get and analyze all sort of data. We'd better use multiple tools to get integrity and accurate result.

VII. FURTHER WORK

Our group will perform with other acquisition and analyzing open source and commercial tools in future.

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Credit Card Fraud Detection Using Online Boosting with Extremely Fast Decision Tree

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Abstract

Nowadays, data stream mining is a very hot and high attention research field due to the real-time industrial applications from different sources are generating amount of data continuously as the streaming style. To process these growing and large data streams, data stream mining, classification algorithms have been proposed. These algorithms have to deal with high processing time and memory costs, class imbalance, overfitting and concept drift and so on. It is sure that ensembles of classifiers are being effectively used to make improvement in the accuracy of single classifiers in either data mining or data stream mining. Thus, to get higher performance in prediction with largely no increasing memory and time costs, this paper proposes an Online Boosting(OLBoost) Approach, which is firstly use the Extremely Fast Decision Tree (EFDT) as base (weak) learner , in order to ensemble them into a single online strong learner. The experiments of the proposed method were carried out for credit card fraud detection domain with the sample benchmark datasets.

Keywords**:** *EFDT, Boosting, Credit Card Fraud, Data Stream Mining*

I. INTRODUCTON

Today, the volume of data generated from different sources is increasing exponentially and the analysis methods of such huge volumes give a competitive advantages for the today's business world. Many applications, information systems used in the modern business organizations are worked with the rapidly changes in environments, in which data are collected in the streaming form, i.e. "Data Streams" (DS). Some obvious such examples are network analysis, traffic control, GPS , mobile device tracking, user's click log mining, credit card fraud detection and so on. In contrast with static data mining, processing of streams need more computational requirements for algorithms to incrementally process incoming examples while using limited memory and time. In addition, by reason of the non-stationary characteristics, prediction models for data streams are regularly also vital to adapt to concept drifts [1].

Data stream mining uses two main analytical methods; classification and clustering. Among many data stream classification learning methods, decision tree learning method is ordinarily used, because of its fast and it can be easily understood. Normally, Decision Tree algorithms can be categorized based on learning style: batch and incremental (online). Likewise of batch data mining, there are data stream classification algorithms and data stream clustering algorithms. Typically, data stream classification is a different alternative of incremental learning classifiers that have to meets for massive streams of data with restrictive processing time, limited memory, and concept drift.

Based on our previous work [22], we have learned that the Very Fast Decision Tree (VFDT) algorithm [3] is one of the famous algorithm in classifying data streams , being an online decision tree with the advantage of a statistical property, the Hoeffding Bound (HB). In the last years, the authors [4, 5, and 6] have proposed a series of modifications to increase the predictive performance of the VFDT algorithm. Strict Very Fast Decision Tree (SVFDT) is proposed in the paper [3] so that to address the memory cost restrictions by keeping the predictive performance,.

Moreover, in the paper [7], the authors introduced the implementation of Hoeffding Anytime Tree- "Extremely Fast Decision Tree" to improve upon Hoeffding Tree (HT) by learning faster and ensuring merging to the asymptotic batch decision tree on a stationary distribution. Furthermore, the authors also concluded that EFDT is good to consider of ensemble by building as a forgetting, decay or subtree replacement approaches in order to deal with the concept drift.

In recent year, it is sure that researchers in data stream mining field are moving their focus to ensemble based learning for various application domains. An ensemble learning method; bagging, boosting and stacking, also called a multiple classifier, is a number of so-called base classifiers (weak learner), in which the prediction results are combined to forecast new received instances. Ensembles have been shown to be a proficient way of improving predictive accuracy or/and being a modeler for complex, difficult learning problem by decomposing the easier sub-problems [1].

In the paper [8], the authors have investigated the use of the SVFDT as base learner for ensemble solutions such as OzaBag, OzaBoost, Leveraging Bagging, Online Accuracy Updated Ensemble (OUAE) and Adaptive Radom Forests (ARF), to reduce memory consumption without harming the predictive performance. The experimental and comparable results are shown using thirteen benchmark datasets.

Thus, inspired by these papers [7 and 8] , we propose the Online Boosting Approach by coupling with the Extremely Fast Decision Tree (EFDT) as base (weak) learner in order to ensemble them into a single online strong learner for credit card fraud detection.

The other remaining sections of this paper are organized as follows. Section 2 discuss how to retrieve information for our survey work of credit card fraud detection. Section 3 describes ensemble approaches for data stream and especially provides online boosting methods while Section 4 discusses the proposed methods in brief. Finally, Section 5 makes the conclusion and identifies the directions for the future work.

II. CREDIT CARD FRAUD DETECTION

Over the past few years, the usage of credit cards is widespread in modern day society. As a result, the fraud of credit card has been kept on growing. However, the major causes of great financial losses is credit card fraud although the careful and responsible usage of credit card provides enormous benefits. In addition, the improvement of new technologies offers additional ways of crime may commit fraud. Actually, Credit card fraud detection is considerably hard, but also prevalent problem to solve [20]. In the literature, many techniques have been proposed to conduct the credit card fraud detection problem. Some researchers use the advanced data mining methods while others use data streams mining in order to detect credit card fraud.

It is certain that credit card frauds can costs customers and banks billions of dollars totally. Thus, the society needs a great and perfect fraud detection system that not only can detect the fraud but also prevent it in advance. Therefore, credit card fraud detection methods need constant innovation and attention.

In addition, some researchers did the parametric comparisons of all the existing data mining technique while some researchers did the literature review of all the work done on fraud detection systems. Many AI and machine learning techniques, data mining, fuzzy logic, sequence alignment, neural network, logistic regression, naïve bayesian, genetic programming, decision tree etc., have developed in detecting for various credit card frauds from massive credit card transactions.

Normally, detection problems are solved with two different styles: batch and online style. In the batch learning style, a model detects occasionally relearnt from scratch (e.g. once a year or month) while the model is updated at once at the time of new data arrival in the online learning style, that credit card transaction in the form of data stream. Therefore, this also subsequently, the detection of fraudulent use by data stream mining is required.

In recent years, many researchers have paid a lot of attention in stream data mining. Thus, streaming data classification and clustering algorithms are discussed with their key features and comparative significance performances. Based upon the recent survey of fraud detection techniques for credit card, this paper mainly focus to present ensemble approach coupling with online(incremental) decision tree technique applied in credit card fraud detection mechanisms in detail.

There are a lot of different credit card datasets with different attributes such as type of fraud, number of fraudulent records and etc., in order to detect fraud. In this paper, we also follow the paper [21] to use dataset obtained from UCSD-FICO competition. It includes 100,000 records of credit card transactions. Each record consists of 20 data fields. The label data as legitimate and fraudulent, already defined by bank is also included. That dataset includes 97% legitimate transactions and the 3% of data records are fraudulent transactions.

III. ENSEMBLES APPROACHES FOR DATA STREAM

For the data stream mining, Ensemble algorithms are most widely used techniques because they can be combined with drift detection algorithms

and include dynamic updates, such as selective removal or addition of classifiers. In paper [9], the authors proposed a taxonomy that is focused on ensemble learning for data streams.

Based on their taxonomy, ensemble techniques for data steams can be grouped as follow:

- Combination – Voting Method / Architecture
- Diversity–Inducer / Building Blocks for Ensemble Solution
- Base Learner – Any stable classifier : Batch / Incremental
- Update Dynamics – how learner takes place in the ensemble manner.

From the paper [9], we have learned the various base learners usage in ensemble approaches and also known that decision trees, especially Hoeffding Tree (HT) and its variations are the most common base learner for ensemble learning in a streaming setting. Likewise, in the paper [1] proposed the grouping of ensemble learning approaches for data streams with respect to the different points of view. The most common categorization are the following:

- stationary vs non-stationary stream classifiers
- active vs passive approaches
- chunk based vs on-line learning modes
- distinguishing different techniques for updating component classifiers and aggregating their predictions.

From these systematic survey, though there are serval not only bagging approaches but also boosting approaches for data streaming, this paper only focus on using the boosting ensemble methods as they offer several recognized guarantees and are mostly effective when the base models are simple.

A. Online Boosting Ensembles

In order to increase the predictive performance of Online Decision Tree (ODTs), a lot techniques have been proposed. Among them the three main groups such as the structural modification of the decision tree the additional prediction strategies with the same structure; and the ensembles approaches are famous [2]. From them, the last, ensembles methods have been many proposed not only bagging and boosting with the online style. Due to the focus of this paper is online boosting, the following paragraph is described some related works.

Oza and Russell introduced [10], a parallel boosting strategy by just following AdaBoost with the exception of weight calculation , because AdaBoost can do training with prior knowledge of the number of instances available and it cannot be use with data stream setting. It is called OzaBoost. Due to its simplicity and efficiency, OzaBoost has also got many great achievement in the real-world applications, especially for the computer vision [11],

Many other online boosting algorithms have been addressed for different application needs, such as semi-supervised learning [11], multi-instance learning [12], and feature selection [13].

Online Gradient Boost [11] is an online variant of Gradient Boost, which uses functional gradient descent to decide the optimal example weights and greedily minimizes the loss function of interest. It is denoted as OGBoost. Online Smooth Boosting (OSBoost) is proposed in paper [14].

In 2004, the paper [2], the authors proposed the Adaptive Boosting ensemble algorithm and did practical experiments with the real life data containing 100 k credit card transactions. In 2013, the paper [14] proposed OzaBoost Dynamic, in which the weight calculation is modified and the number of boosted “weak” learners are used in order to improve its performance in terms of memory consumption and also presented the pragmatic results showing the performance of all algorithms using data sets including fifty and sixty million instances.

In the paper [15], the author proposed an algorithm called Online Non-Stationary Boosting (ONSBoost) that are similar with Online Boosting, and also uses a static ensemble size without generating new members. At each time, new examples are presented and it adapts the changes in the data distribution. Using a parallelizing virtual machine, ONSBoost is evaluated with Online Boosting on the STAGGER dataset and the derived three other datasets.

In the paper [8], the author proposed SVFDT as base learner in not only OzaBoost but also bagging and shown the experimental results. Inspirations based on these literature, this paper propose Online Boosting (OLBoost) Approach by coupling with the Extremely Fast Decision Tree (EFDT) as base (weak) learner and explain detail in the next section.

IV. PROPOSED METHOD- ONLINE BOOSTING WITH EFDT

In 2000, the paper [18] presented Hoeffding Tree, namely Very Fast Decision Tree (VFDT) algorithm. For data stream classification, it is one of the most popular and first algorithms, being capable of

inducing a decision tree in an online fashion with a statistical property, the Hoeffding Bound (HB). Hoeffding bounds is usually applied in the decision based on the number of instances to be trained to reach a certain level of confidence. After that, many variants or modifications of VFDT are emerged and that summary is shown in the table 1, as we described in our previous work [22].

TABLE I. SUMMARY OF ONLINE DECISION TREE

Algorithm	Description
VFDT (2000)	- incrementally learning from huge data streams in a single pass using constant memory per leaf
CVFDT(2001)	- extends VFDT to combine gradual changes in the basic data distribution for concept-drifting
EFDT (2018)	- improves the process of splitting data by letting revision on the split decisions and get great achievements in terms of performance on many datasets
One-Sided Minimum OSMDT (2018)	- Uses local node statistics to optimize the frequency of evaluation of split decisions.
VFDT-variants (2017, 2018)	-improve and enhance the performance of VFDT -to reduce memory cost

The paper [7] has introduced Hoeffding Anytime Tree (HATT) and denoted as "Extremely Fast Decision Tree (EFDT). Hoeffding Tree usually builds a tree increment manner, making the delay in selecting of a split at a node until not only it is confident but also it has identified the best split, and never revisiting that decision. Instead, HATT usually uses to select and split as soon as it reaches confident level and that split is useful, and then revisits that decision, make the replacement with the split if it subsequently becomes evident that split is better. Though EFDT is a learner not for concept drift, the authors [7] observed that it is highly effectively on benchmark dataset and has some inbuilt acceptance to concept drift.

TABLE II. EXPERIMENTAL PLAN FOR PROPOSED METHOD

Ensemble	Base Learner	Dataset
OzaBoost	EFDT(HATT)	Public Dataset for Credit Card Fraud Detection
OGBoost		
OSBoost		
OSBoost.OCP		
OSBoost.EXP		
ONSBoost		
OzaBoostDync		

V. CONCLUSION AND FUTUREWORK

With the aim to reduce costs of loss caused by credit card frauds in every year, this paper proposed an ensemble approach, online boosting using online decision tree, namely EFDT, as base learner. The experimental results of proposed method will be done as our plan and also plan to make comparative study of our proposed method with online boosting using others base learners by using not only UCSD-FICO credit card dataset but also another credit card dataset from Kaggle website.

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Developing and Analysis of Cyber Security Models for Security Operation Center in Myanmar

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Abstract

In counteraction to the increasing threat of cyber terrorism, the modeling to be predicted in guessing the predictive models for estimating the incidence of cyber-attacks for enterprise network in Myanmar are seriously needed. Although we need these models, there is no record of attacks, defenseless, outcome and threat to utilize the developing predictive models and authentication. The main purpose of this research is to determine whether SOC (Security Operation Center) manager uses cyber security model by using SOC results figures to prepare further cyber defense and incident response plan. The goal of this study was achieved by conducting experiments on various cyber-attacks occurred in security operation center of Industrial Control System (ICS).

Keywords: *Blue team, Incident Handling, SOC, Cyber Security Model, Vulnerabilities, Threats, Attack.*

I. INTRODUCTION

Responding to cyber terrorism, the researchers and practitioners need to develop and analyze the cyber security prediction models and to serve as a framework to be used as resource into the scheme to support the decision. Due to the harshness of cyber security problem and the confusion which can be caused by cyber-attacks on the Enterprise's information infrastructure, we are induced to build up the prediction models. The critical factor for cyber threat problems will be expressed in excerpts from the following our SOC reports.

Security analyst which represents the Blue Team of Enterprise network always watching the SOC. As per duties and responsibilities of SOC analyst, there are many events in every times. When he finds the critical events he will report that events to red team and purple team at first [1]. And then he will fix or patch that critical vulnerabilities. After that security analyst had many questions. Which sectors still get vulnerabilities and need to change or manage for coming cyber-attacks? SOC shows events according in their security directive rules. But SOC cannot show which sectors need to change or manage not to damage for upcoming attacks [2]. We will focus beyond the SOC to develop the predictive cyber security model.

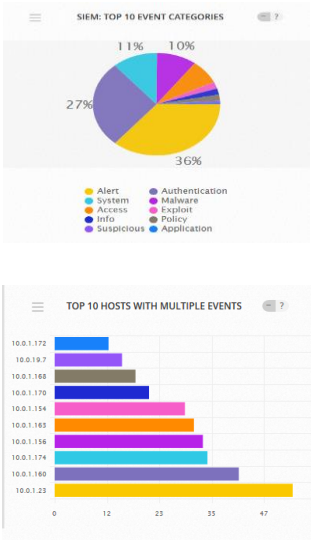

Figure 1. Top 10 events and hosts according SOC

Building up and knowledge of the theory of cyber security models is one of the important matters. Let us mention the types of cyber security models: the risk model and the exponential models. The risk model form one of the models can foresee the threat of attack in a weak position of indicated cyber security situation. It supports the incident handling including networking, log analysis, forensic and packet analysis. The exponential model called the second model, is time based and anticipate the duration between the probability of attacks for a given risk priority. It supports the incident response

including logistic, communications, coordination and planning. The occasion happening during attacks is substitute measure of risks. When duration between attacks becomes longer, the risk of the user is becoming greater because of the result of a growing number of attacks. It is vital to note that we cannot be foreseeable the cyber-attacks which will happen in the near future. So the models mentioned above can be endorsed against the real-world attack. Illustration the important parameters and variables in the security operation center for researchers and SOC managers from Blue team can be only done.

There are the reasons why the experiments were conducted by solving with the help of cyber security. To get the occurring response the OODA Loop method, developed by US Air Force military strategist John Boy is needed to be employed. The OODA Loop method is aimed at attention to the crucial methods to be able to act in response to any information of security crisis: Observe, Orient, Decide, and Act (OODA). SOC can support various observation and orientation. Risk model can do decision making and exponential model can take response actions.

There may be coming up research questions that are crucial in response to cyber security crisis. These concerning questions are as below:

1) Can a variety of presentations and models of cyber security which is advanced provide the frame for researchers and practitioners to promote the field?

2) Can the models of hypothetical predictions be developed to evaluate the threat of different types of cyber-attacks?

Nonetheless, it is still useful to answer the upcoming questions after actions related to potential attacks if we provide explanations, equations, plots and analyses and synthesis. It stands to reason that we need to learn about the components of predictive models and their interactions to reflect objects and events in the enterprise network. The target of these models is for providing the SOC managers while investigating the assumption about how to take actions to cyber-attack prior to occurrence using risks, weakness, duration between attacks and interruption (number and time) concepts.

The rest of this paper is organized as follows. In section 2, we explored related works. Model structure of Risk Model was presented in section 3. Experiment and analysis were mentioned in section 4. We discussed and concluded our research in section 5. We left some future works in section 6.

II. RELATED WORK

Cyber security of Industry Control System (ICS) has become a major topic of active research [3]. It is of great importance and we need to realize the complexity and obstacles with these. They focused a main problem of cyber security research which means that it is supposed to concentrate on occurrence of security in dealing with the information and communication technology [4], while they barely calculate the real result of successful attacks. As a result, Access to risk of quantity is not that easy to get. Some of researches use Preliminary Interdependency Analysis (PIA) models that are well fitted for assessment of quantity of risk, as Both ICT and ICS are widely employed in. Even though PIA had not been used to explicitly express and inform cyber security concerns.

In [5] the researchers extended the risk assessments method Risk = (Asset Value * Event Priority * Event Reliability) / 25. There are several domains within cyber security end users related models, or organizational level modeling, therefore it supposed to narrow down his focus and then start work on it. It also give introduction the perception of cyber security according to its framework, workforces and other related information of personal data in the computer. Their works give results from an efficient audit of time based attacks which ruled with SOC's Host-based Intrusion Detection System (HIDS) and Network-based Intrusion Detection System (NIDS) correlation directives countermeasure. They calculated and correspondence the risk and exponential model under time based attacks. In their study they explore the attack parameters (sensitivity analysis) and time. As a result, the competitor would decide the same even if attacker is addressed at several times the same choice during the violence. That's why studying the behavior of one attacker and attack-strategy at a time will be enough to find out; comparison of the impact of multiple, different classes of intruders and various attack-strategies require building separated models and studies.

In this research [6] cyber security operations center installment of models are proposed to provide better and advancing awareness to situation in order to detect common and frequent advantage, and also approached and cross-channel exploits. 0 (Zero) day exploits are now common and frequent, and impacts far much greater than before. This situation is further made worse by the lack of sufficient and well deployed security operations centers to watch

organizational cyber investments to get perimeter defense.

In this research paper we used Open Source Security Information Management (OSSIM) which was provided AlienVault. It includes HIDS and NIDS.

III. MODEL STRUCTURE OF RISK MODEL

This model relates the basic theory of information security which is probability of attack, likelihood of weak points and outcome of an attack. We assume that risk can be accurately calculated by using equation (1).The justification is that risk in intuitive way would increase as all three quantities comprising equation (1) increase.

Risk (R) = Relative Probability of Attack (A) * Probability of Vulnerability (V) * Consequence (C)

As the symbol, we can assume their identification

$$R_1 = P_{ai} * P_{vi} * C \quad (1)$$

Examples of (2) include:

$R_1 = P_{ai}$ (Malware) * P_{v1} (AD server down) * C_1 (Consequence of Risk priority 1) (2)

$R_2 = R_{a2}$ (Hash dump) * P_{v2} (SMB open) * C_2 (Consequence of Risk Priority 2) (3)

The measure for probability of attack, relative probability of attack, is estimating using the following equation.

$$P_{ai} = \frac{T_L(i)}{\sum_{i=1}^{n} T_L(i)} \quad (4)$$

P_{ai} = Relative probability of risk priority i (attack types).

$T_L(i)$ - Relative threat level of attack of risk priority i

IV. CYBER SECURITY IN ENTERPRISE INFRASTRUCTURE

TABLE I. CYBER SECURITY IN CRITICAL ENTERPRISE INFRASTRUCTURE

Vulnerability V_i	**Consequence** C_i	**Attack Vectors,Risk Priority** A_i
V_1 - Network Firewall	C_1 - Firewall down	A_1 - Denial of Service (DoS)
V_2 -Host firewall	C_2 - AD server down	A_2 - Man in the Middle (MITM)
V_3 - Password Capture Protection (https)	C_3 - Web server down	A_3 - Information disclosure
V_4 - Web Application Firewall (WAF)	C_4 - Operating System crash	A_4 - Malware (Ransomware)
V_5 - Host Malware Protection	C_5 - Application corrupted	A_5 - Virus (damage Database and OS)
V_6 - Host Endpoint Security	C_6 - Hardware Failure	A_6 - Trojan
V_7 - (Network Instruction Detection System (NIDS)	C_7 -Router Misconfiguration	A_7 - Remote Code Execution (RCE)
V_8 - Data Loss Prevention (DLP)	C_8 - Transmission Link Down (ISP down)	-
V_9 - IP security	C_9 - Proxy server down	-
	C_{10} -Power failure	-

R_i - Risk of priority i. Risk priority is the consequence of a given types of attack (DoS, Virus. Malware,etc.)

P_{ai} - Relative probability of risk priority i.

P_{vi} - Probability of vulnerability of risk priority i. we must count or scan this probability in each endpoint.

C_i - Consequence associated with risk priority i. The numbers of network objects that affected (AD Server, WEB server, UPS for power failure, etc.)

$T_L(i)$ - Relative threat level of attack of risk priority i.

N – Total numbers of various attacks.

V. EXPERIMENT AND ANALYSIS OF MODELS

In this section, we calculate Risk Model and Exponential Model. Then analyze Time based attacks and Rate of change of Time between Attack events.

A. Calculation for Risk Model

We will calculate and plot that demonstrate risk model outputs. The sensitive data in the Table I was developed as follows. The first column of the table shows 10 major types of attacks in our Enterprise network. Starting with Alert – the most severe and ending with Application the least severe.T_L (i) also represent a subjective assessments of the relative threat level of various types of attacks,

starting with Alert = 68,700 which was 37% of the whole critical security events, and ending with corrupt of Application = 704 is computed from (4).

Data is not still available for P_{ai} .We calculate the relative threat level (P_{ai}) from real time SOC system. The desired output risk = R_i is computed from Eq (1). The bold values in the table emphasize the significant results. Figure 2 shows how risks differ from the probability of attack. The plot is made notes with the types of attacks associated with the major risk values. As [7] a practical matter, the plot indicates that risk would rise rapidly at a value of $P_{ai} \cong 0.15$.

This could be assumed a major risk and that the blue team should prepare to incident response plan for weak security policy.

TABLE II. ATTACK TYPES AND DATA FOR RISK MODEL

	Attack Type	$T_L(i)$	**Related Threat Level = Relative Probability of Attack** P_{ai}
1	Alert	68,700 (37%)	0.322
2	System	20,615 (11%)	0.096
3	Access	13,687	0.064
4	Info	3,475	0.016
5	Suspicious	1,288	0.006
6	Authenticat ion	49,044 (26%)	0.230
7	Malware	18,230 (10%)	0.085
8	Exploit	3,734	0.017
9	Policy	3,3436	0.157
10	Application	704	0.003
		$\sum_{i=1}^{a} T_L(i)$	$\frac{T_L(i)}{\sum_{i=1}^{a} T_L(i)}$

TABLE III. ATTACKED HOST AND DATA FOR CALCULATION RISK MODEL

	Attacked Host	**Rate of Attack (N(T))**	**Vulnerabilities which is scanning from each endpoint host** P_{vi}	$R_1 = P_{ai} + P_{vi} + C_i$	C_i
1	10.0.1.172	12	0.66	0.21	1
2	10.0.19.7	15	0.55	0.21	4
3	10.0.1.168	18	0.44	0.16	6
4	10.0.1.170	21	0.77	0.03	3
5	10.0.1.154	29	0.55	0.01	2
6	10.0.1.163	31	0.77	0.17	1
7	10.0.1.156	33	0.88	0.15	2
8	10.0.1.174	34	0.33	0.01	2
9	**10.0.1.160**	**41**	0.33	**0.25**	5
10	10.0.1.23	53	0.11	0.01	1

However we note in Table II that R_i is significantly a function of consequences C_i. Thus the analysis of the assignment of C_i to the relative threat levels could be performed.

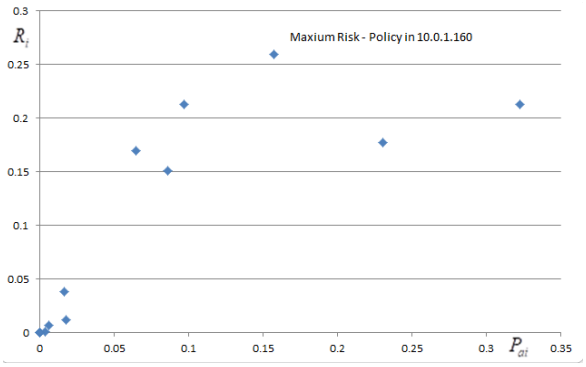

Figure 2. Risk R_i and probability of attack P_{ai}

Figure (1) and (2) shows us the relationships among risk, probability and consequences. We analyzed to have more than one view of the relationships to prove the others. (i.e Figure (3) make proves that the Figure (2) show the Security Policy weakness being the major risks.)

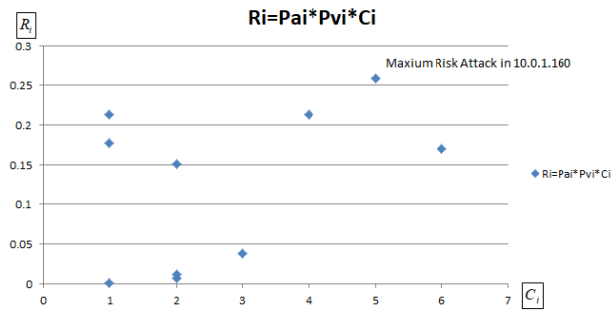

Figure 3. Risk R_i and consequence C_i

Figure (3) shows that risk increases with consequences, as we would expect and again the diagram is annotated with the major risk attacks of Security Policy weakness.

B. Calculation for exponential model

The basic of this predictive cyber security model is that the time between attacks, t_a is a key changeable in working against attacks. P_{ai} is data on probability of attack. We measure relating level of threats, T_L to estimate P_{ai}. t_a can measure the risk because of the smaller the value of t_a which can happen the growing frequency of attack. (i.e. the higher risk). This cyber security model opposed the accounts of risk priority (i) and specific types of attacks and attacked hosts from the events of SOC.

$$P_{ai} = f(T_L) \qquad (5)$$

In the above equation, to develop the model, we formulate probability of attack as a function of relative threat level.

Where, T_L = relative threat level;

t_a = time between attack of type a day.

N (T) = number of attacks in Time T. (we gets that data from our enterprise SOC)

T = specified attack time period in 365 days

$$\lambda = \frac{N(T)}{T} \text{ = rate of attack per day} \quad (6)$$

$$P_{ai} = \lambda e^{-\lambda t} a \quad (7)$$

In the above Eq means the probability of density function for the exponential distribution by assuming for P_{ai}. We calculate the Eq (7) to produce the following equations (8) and (9)

$$\log(P_{ai}) = \log\lambda - \lambda t_a \quad (8)$$

$$\lambda t_a = \log\lambda - \log(P_{ai}) \quad (9)$$

By solving equation (9) for t_a , we obtain equation (10)

$$t_a = \frac{1}{\lambda}\log(\frac{\lambda}{P_{ai}}) \quad (10)$$

C. Analyzing Time Based Attack

The plot of time between attacks t_a and probability of attack P_{ai} is shown in Fig (4). In our events data, there is no saturation $t_a > 24$ hours. Because of low value of t_a imply high frequency of attack.

Growing time of attack may also occur high risk. This security policy would exclude all attacks vectors except Denial of Service (DoS) / Distributed Denial of Services (DDoS). This plot shows us how much we can manage incident handling response as quickly as possible.

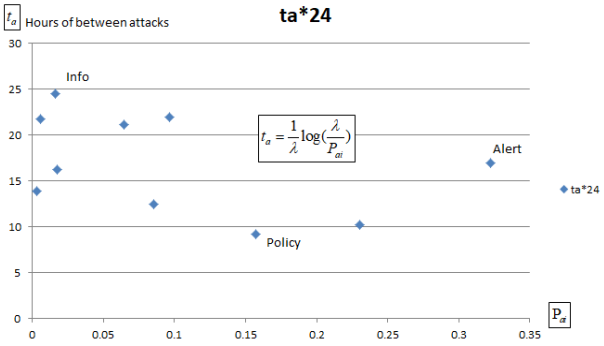

Figure 4. Expected frequency of attack and probable attack $P_a(t)$ at time t 24 hours

If our enterprise network gets DDoS attack. We can measure the increasing traffic based on . By seeing upcoming NetFlow results, Firewall throughputs and DDoS Protectors throughputs, we can countermeasure the stability of DDoS protector live.

TABLE IV. CALCULATION OF TIME BETWEEN ATTACKS

Rates of Attack in 365 days (N(T))	P_{ai}	Attack by Day $\frac{N(T)}{T}$	Days between Attack $\frac{1}{\lambda}\log(\frac{\lambda}{P_{ai}})$	Hours between Attack $(t_a * 24)$	Hours between Attack $\frac{d(t_a)}{d(P_{ai})}$
1200	0.322	3.287671	2.321313915	16.94559158	-0.286727058
1500	0.096	4.109589	3.748187699	21.88941616	-0.611536032
1800	0.064	4.931507	4.340081745	21.12173116	-0.639638551
2100	0.016	5.753425	5.865084423	24.46578073	-1.850949244
2900	0.006	7.945205	7.180361666	21.68964421	-2.618641519
3100	0.230	8.493151	3.607425814	10.19388714	-0.060183666
3300	0.085	9.041096	4.659595437	12.36910789	-0.142880631
3400	0.017	9.315068	6.275036943	16.16744812	-0.657136758
4100	**0.157**	11.23288	4.270095367	9.123423272	-0.050466813
5300	0.003	14.52055	8.387425 3	13.86298974	-1.434377651

D. Rate of change of Time between Attacks events

The rate of change of time between attacks is related to the possibility of attack is obtained by making a distinction (10).

$$\frac{d(t_a)}{d(P_{ai})} = -\frac{1}{\lambda}(\frac{1}{P_{ai}^2})(\frac{P_{ai}}{\lambda}) = -(\frac{1}{\lambda^2})(\frac{1}{P_{ai}}) \quad (11)$$

It gives the (11) which tabulates in Table IV.

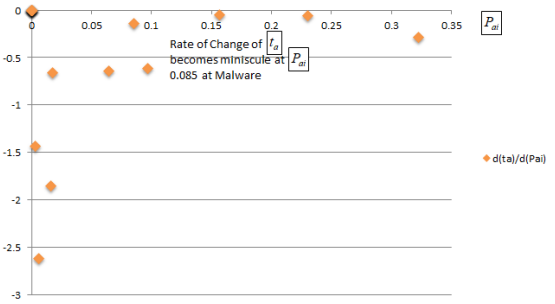

Figure 5. Rate of change of time between attacks $d(t_a) / d(P_a)$ and probability of attacks P_{ai}

That quantity is of interest because we can see when the rate of change of t_a which is a replacement

for risk. t_a becomes small that the threat is virtually nonexistent. That saturation is demonstrated in Fig (5), where P_{ai} = 0.085 corresponds to a Malware attack. At that point, the pace of change is too tiny, meaning that the pace of change of risk is of little.

VI. Discussion and Conclusion

Having stated that time between attacks t_a is a delegate for risk R_i. We analyze and investigate this hypothesis by scheming the former against latter. The result is shown in Fig (6).

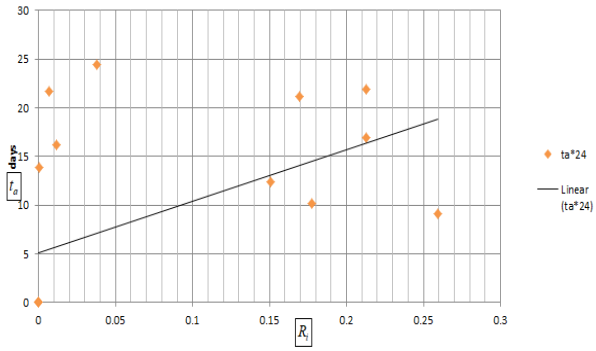

Figure 6. Time between Attacks t_a and Risk R_i

It shows a fairly good correspondence. The fact that the proponent model called exponential in another way is much easier to execute than the risk model is of importance. The former could use the SOC manager's prediction model of choice [8].

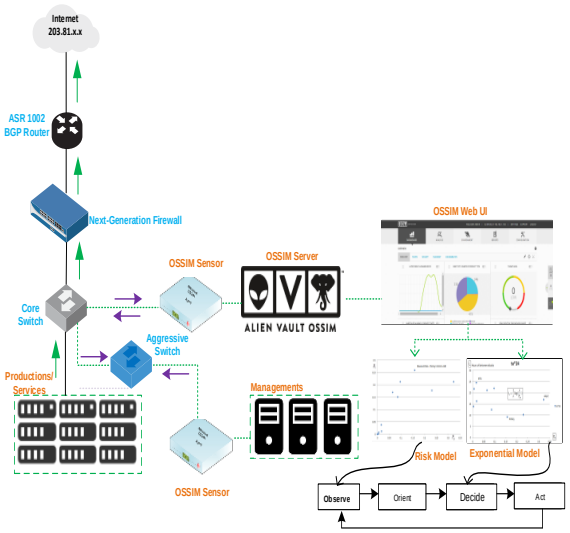

Figure 7. Big picture of SOC analyst for incident handling and response

There are many huge cyber attacks daily in real world. SOC analyst should handle the top major attacks which are pushed by the security policy and best practice for infrastructure security. Risk and Exponential models are fully effectiveness for priority of incident handling and response as OODA loop among that many huge cyber attacks: What is critical and How is it protected ?

In the research question section, we suggested the following questions were answered by this research paper:

1) Can different show off and models of cyber security which is advanced provide the frame for researchers and practitioners to promote the ICT/ICS security research field?

Our paper suggests that, explanation, equations, plots, tables which comprise a frame work depending on our blue team for ICT/ICS security research, the answer is "true".

2) Can the models of hypothetical predictions be developed to evaluate the threat of various and sophisticated types of cyber security attacks?

For answering this question is also "true" we calculate and demonstrate as evidence relevant scheme. (i.e. The Risk Model in Figure-2 identifies the maximum risk attacks, which was security weakness policy. Furthermore, with respect to the Exponential model, Figure-4 provides us with the probability of attack and Time between Attacks threshold.

VII. Future Study

We left some works to do as future research. We are still developing in writing of SOC directive rules to do as future research. Our study is limited to the effect of a malware type of attack on system behavior: a cyber-attack via the Ransomware of a sub-station. According detective of SOC, cyber security model need to extend for multiple stations..

Acknowledgments

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Influence of Voice Delay on Human Perception of Group Synchronization Error for Remote Learning

One-way Communication Case

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Abstract

This paper clarifies the influence of the voice delay on human perception of group synchronization error in remote learning by subjective assessment. In our assessment, we produce the difference in output timing of lecture video between a teacher terminal and a student terminal. Each subject as a student observes the output timing relation (i.e., group synchronization) between the lecture video and teacher's voice, and then the subject answers whether he/she perceives the group synchronization error or not. We also examine the influence of difference in voice contents on the error. Assessment results demonstrate that the perception rate depends on the group synchronization error plus the voice delay, and all the subjects can hardly perceive group synchronization errors (plus voice delays) less than or equal to about 250 ms. We further illustrate that the perception rate depends on the voice contents.

Keywords—*Remote learning, Voice, Video, Group synchronization error, Human perception, Subjective assessment*

I. INTRODUCTION

In multimedia applications such as video conferencing, remote learning, and networked games, the temporal relationships among multiple media streams (for example, voice and video) are important [1]-[3]. Such applications sometimes need to synchronize the output timings of several media streams at all the terminals. If the output timings of each *media unit* (*MU*) [4], which is an information unit such as a video picture and a voice packet for media synchronization, among multiple terminals are different from each other (that is, there exist group synchronization errors), the quality of experience (QoE) [5] may be damaged seriously [6]. This may affect the learning effect in remote learning, for example.

To solve the problem, we need to carry out group (or inter-destination) synchronization control [6]-[10], which adjusts the output timing of media streams among multiple terminals (or destinations). In [11] and [12], two error ranges are employed: One is the *imperceptible range* in which the synchronization error cannot be perceived by users, and the other is the *allowable range* in which the error is felt to be allowable for users. In [15], we carry out the subjective assessment to clarify the allowable and imperceptible ranges of group synchronization error. However, the voice delay is assumed to be negligible small. Also, we handle only a set of voice and video contents. We need to clarify influences of voice delay and voice and video contents on the group synchronization errors. This is because the imperceptible and allowable ranges have not been clarified so far.

Therefore, in this paper, we carry out the subjective assessment of group synchronization error by changing the voice delay to investigate the influence of voice delay in remote learning. We also examine the influence of voice contents on the error.

The rest of this paper is organized as follows. We give an outline of the group synchronization in remote learning in Section 2. We also explain the assessment method in Section 3. Then, assessment results are presented in Section 4. Finally, Section 5 concludes the paper.

II. GROUP SYNCHRONIZATION IN REMOTE LEARNING

In multicast communications, we need to perform group synchronization control, which tries to output each MU simultaneously at all the different terminals. If the control is not carried out, the MU cannot be outputted at the same time at the terminals; that is, the group synchronization error occurs.

The configuration of our remote learning system is shown in Fig. 1. The system consists of N ($\geqq 1$) terminals (one terminal is for a teacher, and the other terminals are for students) [15]. The teacher terminal uses a microphone, and each student

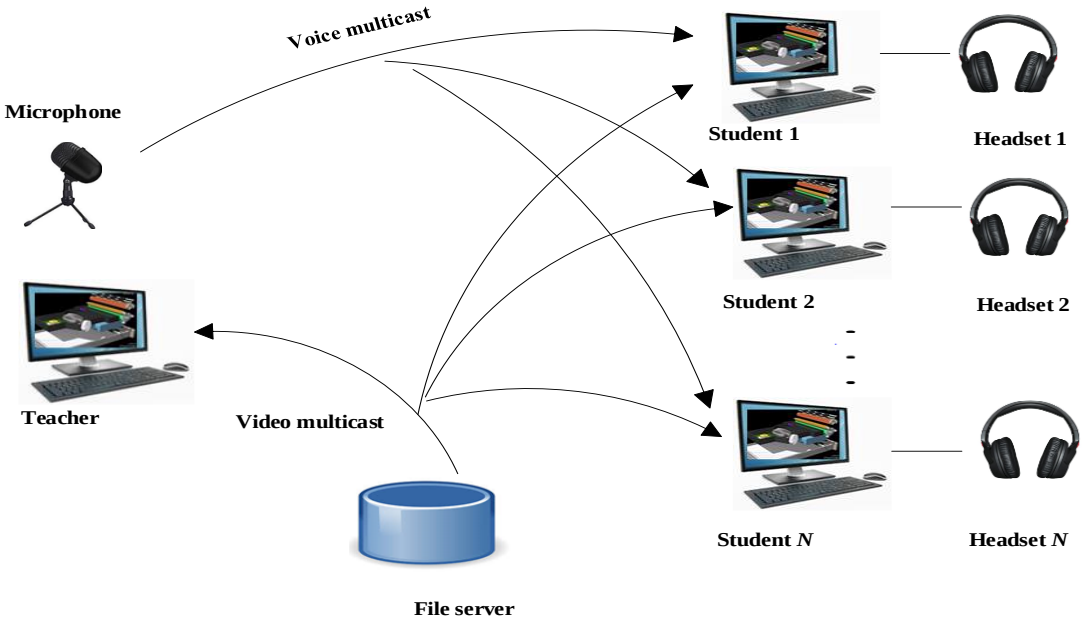

Figure 1. System configuration of remote learning.

terminals employ a headset. The file server multicasts a video stream to all the teacher and student terminals. The teacher orally explains the video contents while watching the video. A voice stream of the teacher captured via the microphone is multicast from the teacher terminal to all the student terminals. Each student listens to the teacher's voice while watching the same video [15].

III. ASSESSMENT METHOD

In the assessment, we set N=1 for simplicity as shown in Fig. 2, where two terminals were used; one was for a teacher, and the other was for a student. For simplicity, the two terminals were located at different places in a room so that the student (teacher) could not see the screen of the teacher (student) terminal. We did not use the microphone at the teacher terminal, but we used the headset for the student terminal.

Alternatively, we saved the teacher's oral explanation as a voice file at the teacher and student terminals, and the corresponding video file was also saved at the two terminals. We produced the group synchronization error of video by changing the start time of video output at the two terminals. The student terminal started to output the voice file after the voice delay when the teacher terminal started to output the voice and video files. By these setups, we constructed our assessment environment without transmitting voice and video between the two terminals. We used the video and voice contents explaining I/O devices such as a mouse and a printer in remote learning, and their output duration was 1 minute and 32 seconds [15].

At the beginning of assessment, we presented a situation in which the group synchronization error is zero and the voice delay is zero to each subject at the student terminal; that is, we started to output the voice and video files simultaneously at the student terminal. We used the single-stimulus method [13].

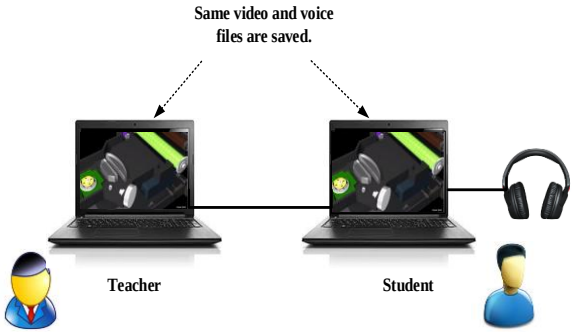

Figure 2. Assessment system

During the assessment, each subject watched the lecture video and listened to the teacher's voice. We changed the group synchronization error from –700 ms to +550 ms at intervals of 50 ms, where negative values denote the teacher ahead of the student and positive values do the teacher behind of the student, and we also changed the voice delay from 0 ms to 100 ms at intervals of 50 ms. Combinations of the error and delay were presented to each subject in random order. To examine the influence of the voice delay more clearly, we also changed the error from –100 ms to 100 ms and the delay from 0 ms to 600 ms at intervals of 50 ms. The subject did not know the value of the error and delay presented in the assessment.

After presenting each combination of the error and delay, we asked the question to each subject (student) as follows: "Did you perceive the group synchronization error?" The subject answered either "Yes" or "No." He/she judged whether the error was perceived or not by monitoring the temporal relation between the teacher's voice and displayed video images.

The total assessment time per subject was about 95 minutes including the explanation time of how to judge the error. The number of subjects (students) was 15 females, and their ages were between 28 and 37.

IV. ASSESSMENT RESULTS

We plot the *perception rate* which is here defined as the percentage of subjects who perceived group synchronization errors as a function of the group synchronization error in Fig. 3, where we set the voice delay is 0 ms, 50 ms, and 100 ms. We also show the perception rate versus the voice delay for group synchronization errors from –100 ms to 100 ms in Fig. 4.

In Fig. 3, we see that the perception rate is 0% when the group synchronization error is between around –250 ms and 250 ms and the voice delay value is 0 ms. As the absolute error exceeds about 250 ms, the perception rate starts to increase; when the absolute error is larger than or equal to about 500 ms, the perception rate is 100%. We also find in the figure that when the delay is 50 ms and 100 ms, we can obtain the perception rate by shifting the results at the delay of 0 ms leftward (i.e., the negative direction) by 50 ms and 100 ms, respectively.

In Fig. 4, we notice that the perception rate starts to increase when the voice delay exceeds about 250 ms and the group synchronization error is 0 ms; for the other errors, we can obtain the perception rates by shifting the rate at the error of 0 ms by the error. For example, the rate at the error of 100 ms is gotten by shifting that at the error of 0 ms leftward by 100 ms.

According to the above discussions, we may be able to say that the perception rate depends on the group synchronization error plus the voice delay. To confirm this, we plot the perception rate versus the error + delay in Fig. 5, which includes the perception rate versus the error (the delay: 0 ms) for different voice contents from those in this paper (called the latter *Voice 1* and the former *Voice 2* here. We will explain the results later). The figure reveals that the perception rate is almost the same as that at the voice delay of 0 ms in Fig. 3. Therefore, we can conclude that the perception rate depends on the group synchronization error + the voice delay.

From Fig. 5, we can also know the imperceptible range and the allowable range. If we assume that the imperceptible range is a range in which the perception rate is less than 20%, the range is between around –250 ms and 250 ms. If the allowable range is assumed to be a range in which the perception rate is greater than or equal to 60% (the value of 60% is just an example), the range of the absolute error is larger than about 300ms.

Furthermore, in Fig. 5, the shape of the perception rate seems to be almost line-symmetric. To confirm the line-symmetric property, we checked whether there are significant differences between the positive part and the negative part in the figure by carrying out t-test. As a result, the p-value were greater than 0.05 [14]. Therefore, we can say that the line symmetry property exists in the figure.

In addition, to examine the influence of difference in voice contents, we show the perception rate in Fig. 5 when we used the voice contents in [15] (i.e., Voice 2) and the voice delay was 0 ms. Note that Voice 2 also explains the video contents handled in this paper. The number of words in Voice 1 is about 110, and that in Voice 2 is around 170; that is, Voice 1 is simplified from Voice 2 by reducing the number of words, and Voice 1 has longer silence periods than Voice 2. From Fig. 5, we notice that the perception rate for Voice 2 is 0% when the group synchronization

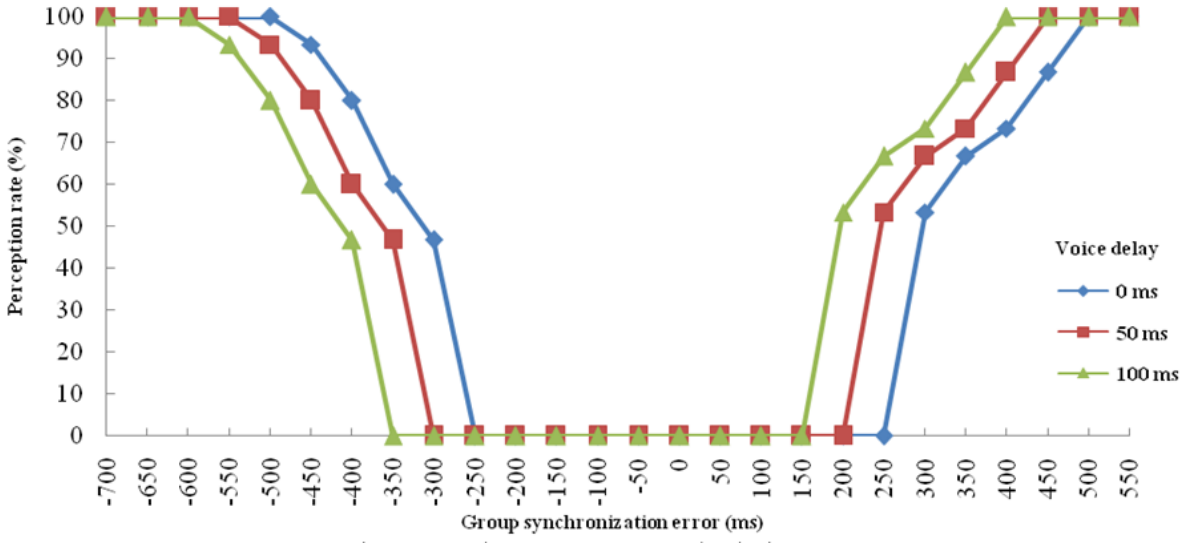

Figure 3. Perception rate versus group synchronization error.

error is between –150 ms and 150 ms (these results are almost the same as those in [15]); as we described earlier, the perception rate for Voice 1 is 0% when the error is between –250 ms and 250 ms. That is, the imperceptible range of Voice 1 is different from that of Voice 2 (on the other hand, the allowable ranges of Voices 1 and 2 are almost the same as each other). Therefore, we can conclude that the perception rate of group synchronization error depends on the voice contents. We need to clarify the influence of the voice and video contents on the perception rate; this is for further study.

V. CONCLUSION

In this paper, we investigated the influence of voice delay on human perception of group synchronization error for remote learning by carry out subjective assessment. Assessment results showed that the perception rate depends on the group synchronization error + the voice delay, and the perception rate is 0% when the error + delay is from about –250 ms to 250 ms. We also confirmed that the perception rate is dependent on the voice contents.

As our future studies, we will carry out the assessment with other video and voice contents. We will investigate the two-way communication case in which a teacher and students can interactively discuss the lecture.

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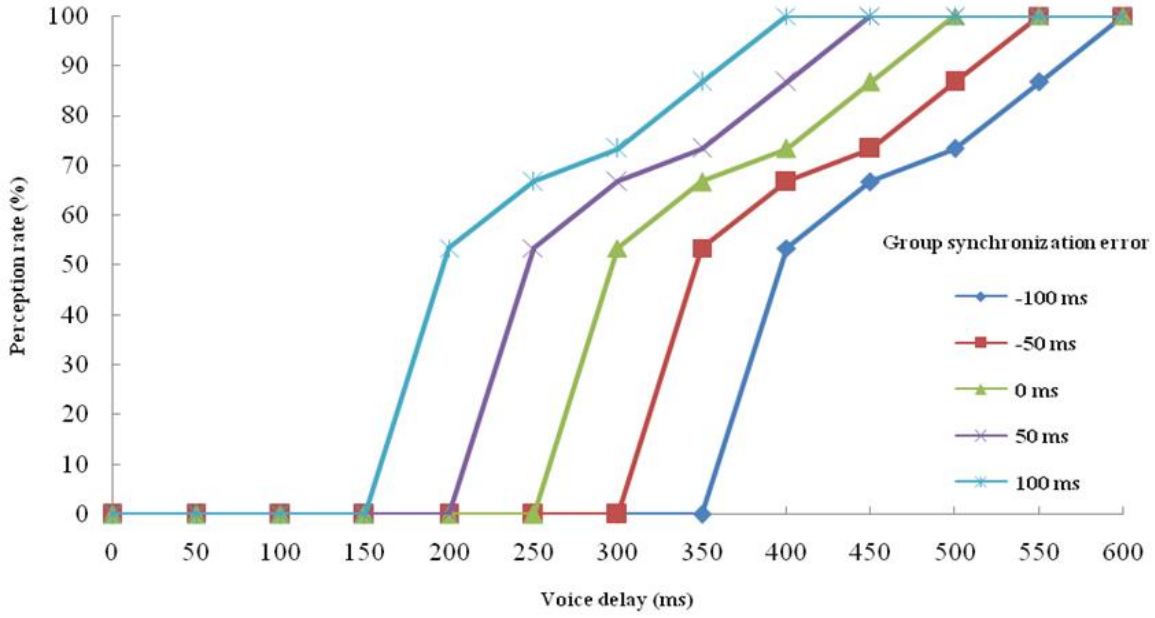

Figure 4. Perception rate versus voice delay.

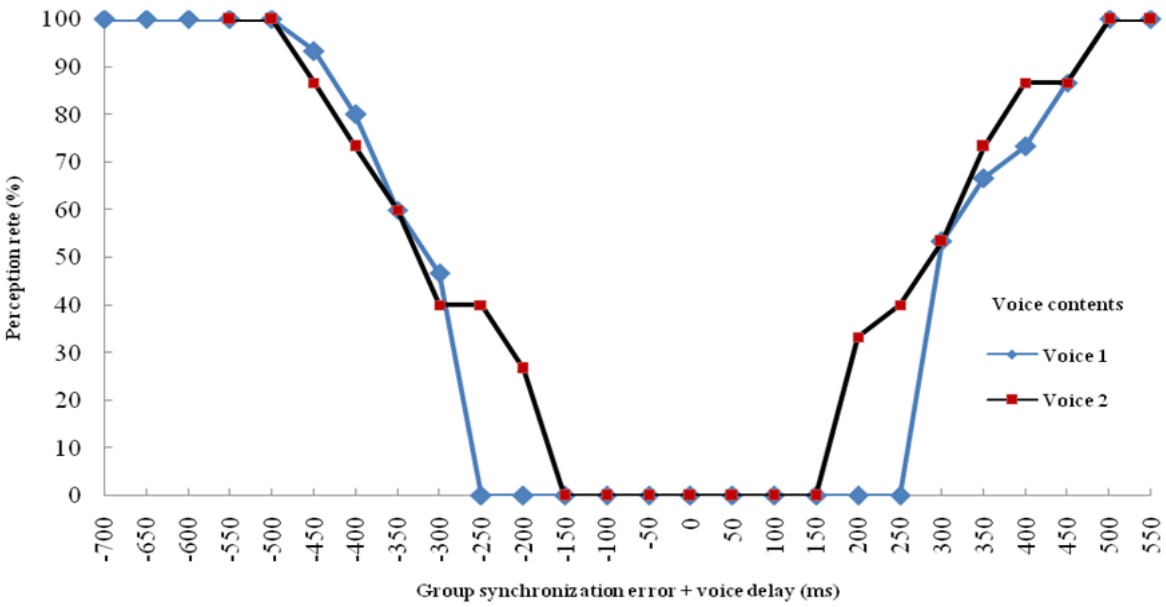

Figure 5. Perception rate versus group synchronization error + voice delay.

IoT Botnet Detection Mechanism Based on UDP Protocol

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Abstract

Today is the time of the Internet of Things (IoT), a great many devices, for example, smart homes, smart retail, smart phone identification, smart lighting, and so forth are being associated with the Internet. There are different devices that are interconnected to a different device on the Internet of things that offer various procedures and forms. The Forensic specialist will have many difficulties to look into gathering the bit of proof from the tainted segment on the IoT devices and furthermore will confront complexities to break down those proof. This paper introduces a UDP flood attack begins by sending countless UDP packet from various IP addresses. The graphical proof is likewise displayed for the DDOS attack utilizing UDP packet flooding. We will do the network forensics investigation for flooding attacks on IoT environments Using Wireshark

Keywords: *Internet of Things, IoT Forensics, Botnet, DoS, DDoS*

I. INTRODUCTION

The Internet-of-Things (IoT) is developing quickly, making openings and difficulties for investigators of a crime, including cyberattacks and physical ambushes (Kebande, 2017, Akatyev and James, 2017). By definition and configuration, keen homes and other IoT situations are associated, dynamic, and can be changed from anyplace whenever (Minerva, 2015; Loung, 2018; Barnard-Wills, 2014). Numerous IoT gadgets have sensors or actuators that produce information, now and again independently and at times in light of human activities (movement discovery, entryway opening). This constantly dynamic, continually producing makes them astounding computerized observers, catching hints of exercises of potential use in examinations. IoT gadgets can be significant wellsprings of proof gave computerized investigators can deal with the amount of information created, the number and assortment of gadgets, the heterogeneity of conventions utilized, and their dispersed nature [1]. A DDoS Attack is one of the most well-known and significant risks to the Internet in which the objective of the attacker is to devour PC assets of the person in question, generally by utilizing numerous PCs to send a high volume of apparently authentic traffic mentioning a few administrations from the person in question. Accordingly, it makes arrange blockage on the target, along these lines disturbing its typical Internet activity.

The transport layer gives a mechanism to the trading of information between end frameworks. Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) are two primary transport protocols that give connection-oriented and connectionless administrations individually. TCP guarantees dependable and requested information conveyance while additionally presenting handling overhead and bandwidth constraints because of congestion and flow control mechanisms. The lightweight UDP neither gives solid conveyance nor experiences preparing overhead and bandwidth confinements and subsequently is utilized in time-sensitive applications on the grounds that dropping

packets is desirable over hanging tight for postponed packets, which may not be an alternative in a constant framework like Voice over IP (VoIP), IPTV, video on demand and web-based gaming [2]. Specifically, a UDP flood attack happens when an attacker creates various bundles to arbitrary goal ports on the unfortunate victim's computer. The unfortunate victim system, on receipt of the UDP packet demands, would react with proper ICMP bundles, if the port is shut. An enormous number of packet reactions would hinder the framework or crash.

II. Related Work

Ryba in [3] in detail depicted the state of the art of research suggestion for counteracting, distinguishing, and the following upgrade and dispersed reflected renouncing of organization assaults similarly as investigated boundary frameworks against the source an IP address spoof, which is major for the increase and the DRDoS assaults.

It is also imperative to suggest the paper of Bekeneva in [4], where the tests DRDoS assaults and security frameworks against them are presented.

IoT devices and DRDoS assaults. A couple of investigators have inspected the DRDoS assault, in any case, only two or three they have focused on IoT devices. Generally, the investigators endeavored to portray the condition around Mirai botnet and to set up a specific proposition for the endorsed methodology or for lively security models for IoT contraptions, and wholesalers [5, 6].

Correspondingly, as in the past subject, there is moreover a push to find numerical or amusement models for DRDoS assault reliant on IoT contraptions and their consequences for sorting out security, for instance [7].

III. Internet of Things (IoT)

The Internet of Things (IoT) depicts the arrangement of physical items—"things"— that is introduced with sensors, software, and various advancements to partner and exchanging data with various gadgets and frameworks over the internet. These gadgets go from customary nuclear family articles to complex mechanical contraptions. With more than 7 billion related IoT gadgets today, masters are envisioning that this number ought to create to 10 billion by 2020 and 22 billion by 2025 [8]. In the course of recent years, IoT has gotten one of the most significant advances of the 21st century. Since we can interface ordinary objects—kitchen apparatuses, vehicles, indoor regulators, child screens—to the internet by means of embended devices, consistent correspondence is conceivable between individuals, procedures, and things.

While the possibility of IoT has been in presence for quite a while, an assortment of ongoing advances in various advances has made it down to earth.

- Access to minimal effort, low-control sensor innovation.
- Connectivity.
- Cloud figuring stages
- Machine learning and investigation.
- Conversational computerized reasoning.

Industrial IoT (IIoT) alludes to the usage of IoT development in industrial settings, especially for instrumentation and control of sensors and gadgets that interface with cloud advances. Starting late, businesses have used machine-to-machine correspondence (M2M) to achieve remote automation and control. In any case, with the improvement of cloud and joined progressions, (for instance, examination and AI), adventures can achieve another computerization layer and it makes new salaries and strategies. IIoT is now and again called the fourth influx of the industrial unrest, or Industry 4.0. Coming up next are some essential uses for IIoT are Smart, assembling, Preventive and prescient support, Smart power networks, Smart urban networks, Connected and shrewd collaborations and Smart mechanized inventory chains.

A. IoT Lifecycle

An IoT system is involved in associated devices that are much of the time sending information about their status and condition around them.

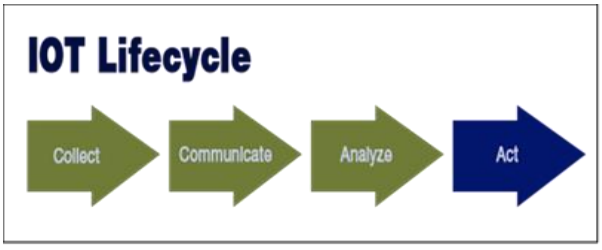

Figure 1. IoT Lifecycle

Collect: The existence cycle of IoT begins with gathering information from various sources conveyed in a specific district. These sources could be any sensors or devices equipped for transmitting information associated with a portal. Information is

productively gathered and gone ahead through a correspondence channel for investigation.

Communicate: This stage includes the protected and solid exchange of information. Switches, switches and firewall advancements assume a crucial job in setting up the correspondence between devices. The Information is sent to the cloud or other server farms utilizing the internet which is our significant methods for correspondence in IoT.

Analysis: This stage is a significant piece of the IoT lifecycle. In this stage information gathered from the various sensors, devices are gathered and examined dependent on the utilization case to separate some valuable yield/data.

Action: This is the last phase of the IoT lifecycle. Data got by the investigation of sensor information is followed up on and appropriate moves and measures are made dependent on the examination result [9].

B. IoT Forensics

IoT devices have limitations in battery, calculation, memory, and radio data transfer capacity. Along these lines, applying security arrangements that for the most part requires overwhelming correspondence burden and more calculation assets, are difficult. Validation, get to control and malware recognition of helpless IoT devices should be considered. The IoT including devices, service, and networks are defenseless against various attacks, for example, physical, software, DoS, DDoS, sticking, spoofing, man-in-the-center and protection spillage. Most IoT security dangers originate from uncertain IoT devices, the attacker focus to exhaust the compromised IoT devices asset particularly network traffic. Network forensics includes catching account and breaking down of network traffic. Serves to gather data, proof assembling and identify attacks. The procedure of examination happened in the network with dealing with the traffic and action. Not quite the same as the other technique, the network forensics-identified with dynamic data that effect is lost.

C. Forensic Investigation in IoT Environment

The Internet of Things (IoT) represents various novel and convoluted difficulties in the field of advanced crime scene investigation. Assessments express that the quantity of arranged devices will remain at 50 billion by 2020, and said devices will create a significant measure of information (Botta , 2014). The handling of gigantic measures of IoT information will prompt a proportionate ascent in the remaining tasks at hand borne by server farms; this will, thus, imply that suppliers are left to manage new provokes identified with limit, security, and investigation.

Guaranteeing that said information is dealt with advantageously comprises a significant test, since the application execution, in general, depends intensely on the information the board administrator's properties (MacDermott, 2018). It is felt that IoT criminology comprises of a blend of three advanced legal sciences plans: cloud level forensics, device-level forensics, and system level forensics (Zawoad and Hasan, 2015) as appeared in figure 2.

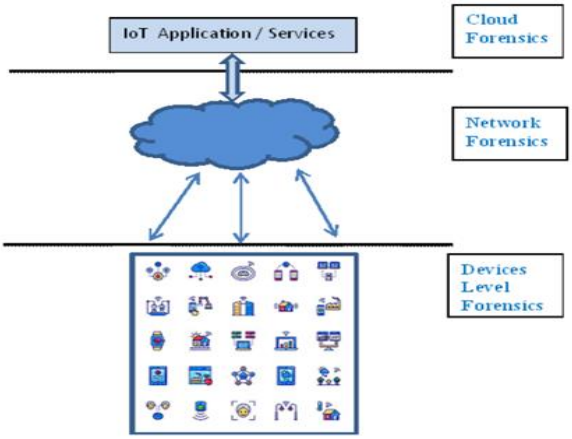

Figure 2. IoT Forensics

Device-level forensics: At this level, a criminological agent needs to gather information first from the nearby memory contained in the IoT device to be dissected. It is important to utilize the IoT device that is missed in breaking down information on the criminological level device.

Network-level forensics: To recognize different sources of attacks can be distinguished from network traffic logs. Hence, the log traffic network can be critical to deciding the blame or opportunity of the suspect. IoT infrastructure incorporates different types of networks, for example, Body Area Networks (BAN), Personal Area Networks (PAN), Home /Hospital Area Networks (HAN), Local Area Networks (LAN) and Wide Area Networks (WAN). Significant proof acquired is gathered from one of these networks with the goal that network forensics.

Cloud level forensics: Cloud forensics is one of the most significant pieces of the IoT scientific space. Why? Because of the way that most existing IoT devices have a low stockpiling and registering limit, information created from IoT devices and IoT networks are put away and handled in the cloud. This is on the grounds that cloud solvents offer an assortment of points of interest including comfort,

enormous limit, adaptability, and availability on demand [1].

IV. BOTNET

In this segment, data identified with Botnets, Botnet Environment, architecture, and activities are given.

A. Botnet Environment

Botnets have had a rich history and movement reliably, defiling and upsetting PC and framework structures. From the outset, botnets were made for caring purposes, with their basic limit being to give credible help to Internet Relay Chats (IRC), a kind of correspondence acclaimed during the '90s. The first IRC bot appeared in 1993, was named Eggdrop and offered help to IRC bot appeared in 1993, was named Eggdrop and offered help to IRC channel correspondence. Following Egg drop, the first perilous bots appeared, with GTbot in 1998 being the first of its sort, which had the choice to execute substance when influenced through its Command and Control (C&C) IRC channel. For example, different bots. It was brought down in December 2009. Another obvious achievement for botnets in 2009 was the proximity of the ancestor of preservationist botnets, where botnets use telephones as their bots (zombies), named SymbOS \ Y xes which centered Symbian contraptions and utilized SMS messages to self-duplicate. Following the surfacing of SymbOS, the first botnet concentrating on Android contraptions named Geinimi was seen, during the completion of 2010. Basically found in China, it utilized a brief HTTP-based C&C structure and was set prepared for sending SMS, messages, bring the zone of the undermined contraption and also made conceivable the further spread of malware.

Generally, botnet makers have manhandled the wide confirmation and strong widening of the IoT, and we need to begin at now scene instances of IoT botnets and what they are set up to do. Botnets included IoT contraptions were the going with the formative improvement of botnets. The most outstanding first appeared in September 2016, under was related as Mirai. Mirai played out probably the most overwhelming DDoS attacks in Internet History, explicitly: 620 Gbps against Brian Kreb's site page, 1.1 Tbps against French Cloud authority affiliation OVH and in October 2016 ambushed Dyn ace concentration and separate down bits of the web like Twitter, Netflix and GitHub. After the nearness of Mirai's source code, different assortments showed up Persirai which is dynamic since, After the nearness of Mirai's source code, different assortments showed up Persirai which is dynamic since April 2017, a more refined understanding of Mirai which targets specific devices of select sellers. Other IoT botnets unite Hajime, which appeared in October 2016, and utilized a decentralized C&C framework that appeared to 'shield' contraptions from Mirai ailments. At long last, BrickerBot was found in April 2017, and as the name proposes attempted to 'square' IoT contraptions in what can be viewed as a permeant DoS assault [10].

B. Botnet Architectures and Characteristics

Botnet models join a few sections. Notwithstanding, a bot is a program that, in the wake of landing at a vulnerable host, sullies it and makes it a pinch of the Botnet. Bots change from other malware, in that they join a channel of correspondence with their makers, empowering them to offer commands to their arrangement of bots (i.e., zombies) and thusly making botnets flexible concerning their handiness. A botnet's malware gets given to frail fixations through what is known as a spread instrument. Most usually there exist two sorts of development, saved and dynamic. Torpid growth strategies predict that clients should locate a functional pace, or other exchanged off-sort out portions and through client affiliation download the malware (bot), dirtying it and making it part of the botnet. Dynamic or self-development methodologies use sub-segments of their framework to effectively check the Internet for uncovered devices, endeavoring to mishandle the identified vulnerabilities, changing the undermined hosts into bots themselves.

The trademark that makes botnets fascinating is where that they permit their controller, by and large, suggested as a botmaster to offer orientation to their arrangement of spoiled devices and get a responsibility, as appeared in Figure 3.This is made conceivable through a Command and Control (C&C) structure. There exist different sorts of C&C frameworks subject to their topology and those sorts are: bound together, P2P, dynamic and crossbreed. In a consolidated topology, bots accomplice, get rules and report/pass on their work in the focal establishment, with most essential developments used here being IRC and HTTP shows. The basic weight of the bound together

topology is that the C&C is a singular inspiration driving disillusionment.

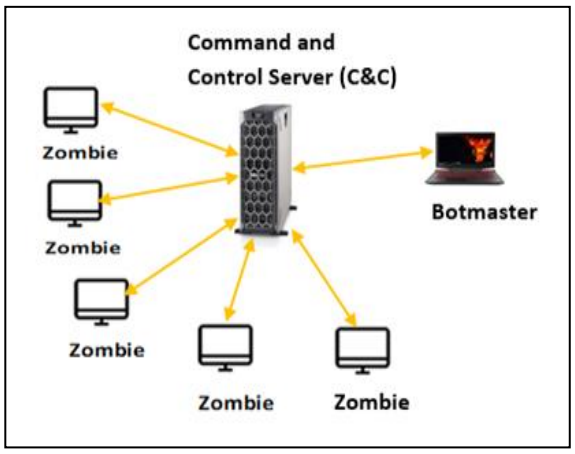

Figure 3. Centralized Botnet and activities

Finally, right now, botmaster fuses center individual bots between their machine and the botnet, with each bot sending commands to the bots that they wrangled, making an other leveled topology and making takedown endeavors difficult, correspondingly as allowing the botmaster to rent bits of their botnet [10].

C. Botnet Activities

Botnets are clearly the most versatile bits of code to explore the Internet. The standard inspiration driving why they get so a great deal of thought isn't an inevitable result of the wonderful ways that botmasters use to scatter their bots from law basic, yet rather the pleasing farthest arrives at that botnets have and the affiliations they oblige the botmasters and their clients. There are varying hacking frameworks used by botnets, including Distributed Denial of Service ambushes (DDoS), Keylogging, Phishing, Spamming, Snap mutilation, Click duplicity and even the enlargement of other Bot malware [10].

V. WHAT IS DDOS ATTACK?

A Denial of Service (DoS) attack is an endeavor by an attacker to make organize assets inert to its real clients by flooding the service's host. Distributed Denial of Service (DDoS) attack is a DoS attack that is begun from various sources. By and large, DoS attack is started from one gadget or virtual machine utilizing Internet association while DDoS attacks are started from a wide range of compromised devices, virtual machines to over-burden the victim frameworks. DDoS is performed by sending an extensive number of solicitations all the while through botnets and compromised IoT devices to exhaust registering assets (Bandwidth and Traffic) of the objective. The compromised devices which are likewise called bot or Zombie works under the supervision of one or huge numbers of the bot-masters and attack controls gatherings of bots (botnet) remotely as in Figure 4. Bots can be either malicious clients whose expectation is an attack or authentic clients who are contaminated.

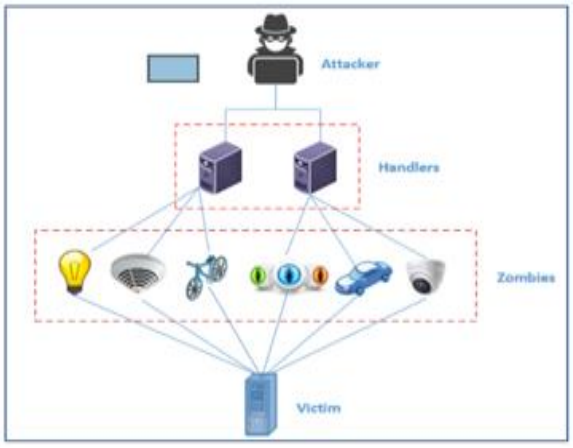

Figure 4. DDoS attack network infrastructure.

A. Direct and Indirect DDoS attack

The DDoS attack can be launched in two different ways either legitimately or with a reflector as in Figure 5. In the immediate system attack, the attackers legitimately send the packets to the objective victim machine. Notwithstanding, an aberrant attack which is likewise called enhancement or reflection attack the attacker utilizes a reflector server and the attacker spoofs the source IP. The attacker sends the IP packet to the reflector server, and afterward, the reflector server sends the reaction to the objective. In the immediate attack, the victim gets the packet with a similar payload as sent by the attacker while in a circuitous attack the reflection server enhances the solicitation it gets from the attacker and sends the reaction to the victim.

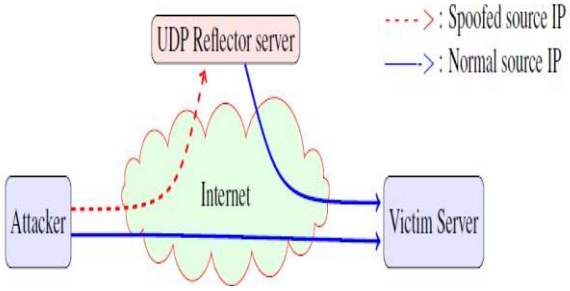

Figure 5. Direct and Indirect Attack

Begin to frame a DDoS attack, at first, attackers recognize vulnerabilities of one or various gatherings of IoT devices to introduce malicious software on them. At the point when malicious software is introduced on the devices, they are called zombies. At that point, the attacker's structure an enormous gathering of zombies geologically distributed which are known as botnet. Each gathering of zombies has a handler which is a software bundle set over the Internet. The handlers are legitimately speaking with attackers and zombies since they have data about the dynamic zombies. While propelling an attack, attackers send the attack to the zombie handlers who will disseminate the attack to all zombies. At that point, zombies will attack the objective framework. DDoS attacks which are created by spoof IP is trying to deal with and channel [11].

B. IP Address spoofing in DDoS attacks

IP address spoofing is utilized for two reasons in DDoS attacks: to Masking botnet devices zones and to arrange a Reflected DDoS.

Masking botnet devices: A botnet is a social event of malware-contaminated gadgets remotely constrained by blameworthy gatherings without the data on their proprietors. They can be encouraged to everything looked at entryways as a given district or server, equipping liable gatherings with the arrangement and frameworks organization preferences for delivering colossal traffic floods. Such floods connect with botnet managers, (a.k.a. shepherds), to help their objective's preferred position limit, accomplishing server singular time and framework immersion. Botnets are generally included either discretionary, topographically dispersed gadgets or PCs having a spot with a similar exchanged off framework (e.g., hacked encouraging stage). By utilizing derided IP passes on to shroud the genuine characters of their botnet gadgets, blameworthy gatherings plan to: Avoid revelation and suggestion by law need and legal automated bosses. Keep bases on lighting up contraption proprietors about an assault in which they are accidentally taking an interest. Avoid security substances, gadgets, and organizations that endeavor to moderate DDoS attacks through the boycotting of assaulting IP addresses.

Reflected DDoS: A reflected DDoS attack uses IP parodying to make fake sales, clearly to support a goal, to move responses from under-verified center individual servers. The's guilty party will presumably improve its traffic yield by enacting huge responses from a ton of more diminutive sales. Fundamental reflected DDoS attack systems include:

- *DNS amplification* – An ANY request beginning from an objective's satirize address is sent to various unbound DNS resolvers. Every 60-bytes deals can incite a 4000-bytes reaction, drawing in assailants to increase traffic yield by as much as 1:70.
- *Smurf attack* – An ICMP Echo request is sent from a goal's satirize address to a widely appealing convey arrange, actuating answers from every device on that system. The degree of amplification relies upon the number of devices to which the sales are imparted. For example, a system with 50 related hosts realizes a 1:50 amplification.
- *NTP amplification* – A get monist request, containing a goal's parodied IP address, is sent to an unbound NTP server. As in DNS amplification, a little sales trigger much greater response, allowing the best amplification extent of 1:200. for how the ridiculed IP is delivered in DDoS attack[11].

VI. WHAT IS A UDP FLOOD ATTACK?

UDP flood is a kind of Denial of Service (DoS) attack in which the assailant overwhelms random ports on the concentrated on the host with IP parcels containing UDP datagrams. The getting host checks for applications identified with these datagrams and—finding none—sends back a "Goal Unreachable" bundle. As progressively more UDP bundles are gotten and answered, the system gets overwhelms and dormant to various clients. Some working framework avoids the UDP flood by constraining the amount of ICMP response [12].

A. UDP Flood DDoS Attack Scenarios

In this scenario uses to perform forensic testing of the IoT device in recognizing flooding attacks using Wireshark. The chose dataset is "IoT_Dataset_UDP_DDoS__00001_20180604180103" in Bot-IoT Dataset was utilized for network forensic in UDP DDoS flooding attack.

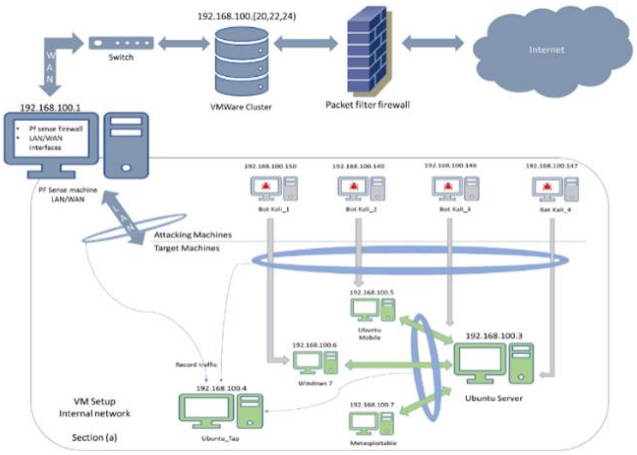

Figure 6. BoT-IoT Dataset

Cyber Range Lab of The focal point of UNSW Canberra Cyber, as appeared in Figure 6. The environment consolidates a mix of typical and botnet traffic. The dataset's source files are given in various formats, including the first pcap files, the produced argus files, and CSV files. The files were isolated, in view of assault classification and subcategory, to all the more likely aid the naming procedure.

The caught pcap files are 69.3 GB in size, with more than 72.000.000 records. The extracted flow traffic, in CSV format, is 16.7 GB in size. To facilitate the handling of the dataset, we extracted 202.7MB of the original dataset is 13.7GB of UDP_DDOS pcap files.

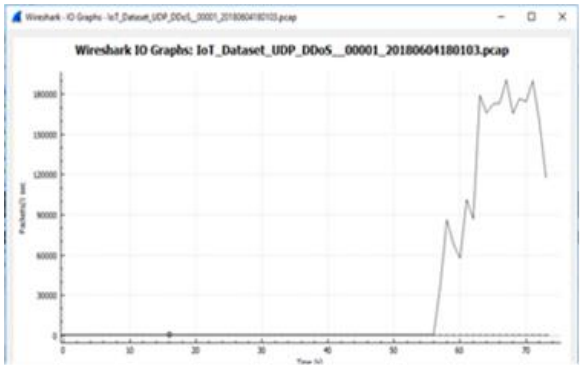

Figure 7. IO Graph for IoT_Dataset_ UDP_DDOS pcap file

Flooding attacks will be visible when the request to the IoT device increased capture traffic that is an anomaly. Then flooding attacks are sent from the attacker so that traffic will increase.

Figure 8. Traffic Log in Wireshark

After the log files are recorded, the log file will be taken and analyzed using Wireshark to have this forensic evidence.

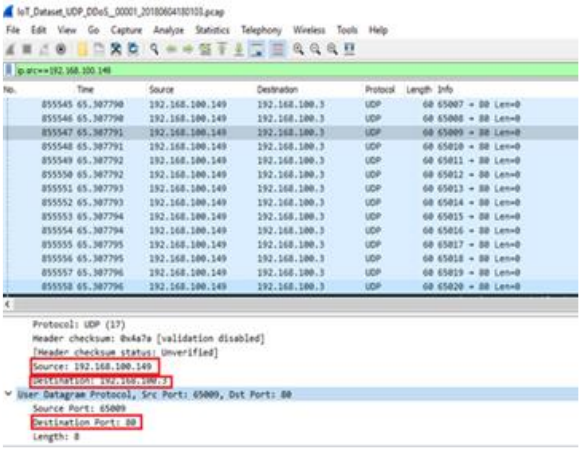

Figure 9. UDP Flood packet being sent to port 80

Above figure, the server IP is 192.168.100.149 and it send UDP packets to 192.168.100.3 with port 80. This is profoundly unusual and as a rule, UDP does not have to send to port 80 genuinely. These are the first signs of a UDP flood attack.

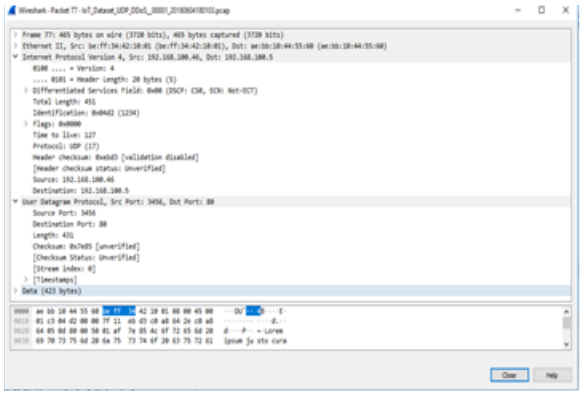

Figure 10. UDP Follow

The data got from the proof follows is utilized to distinguish the episode. This will help in source trace back, reproduction of the assault situation and attribution to a source. From the collection of the line can have one line to perform analysis on any part of the frame that represents a frame in an attack packet flooding of IP address 192.168.100.46 has a length (length) range in the 465 Bytes). On the Internet Protocol Version 4, to read as 192.168.100.46 IP source and destination IP address visible 192.168.100.5 with 20 Bytes header length and the total length of 451. On the part of the user datagram protocol, source port reads as 3456 and destination port read as 80.

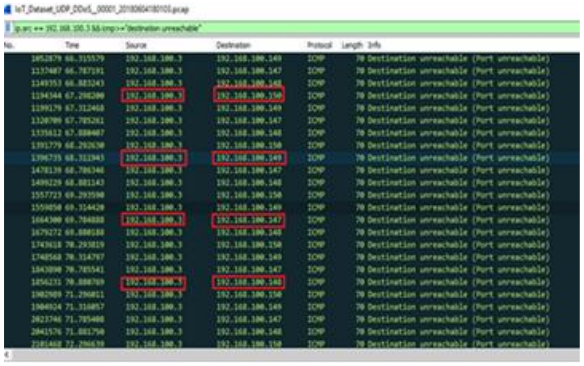

Figure 11. Detecting ICMP host unreachable packet

Server IP is 192.168.100.3. The server is sent to 192.168.100.150, 149,147148 ICMP host unreachable packet. So, this four IP address would be the victim IP.

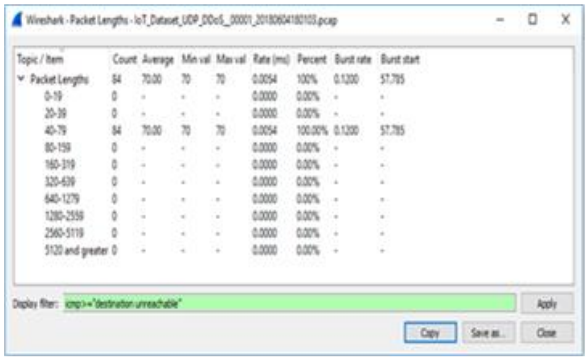

Figure 12. Packet Lengths for Destination Unreachable

ICMP destination unreachable packet number is 84 and Capture traffic increased in 40-79 packet length. We noted the following:

- If no service is listening on that UDP port, the server responds to the client with an “ICMP host unreachable” packet.
- Thus there is a high chance of being this DDoS UDP flood attack.
- Logfile data with p.cap expansion can be broke down by network forensic investigation utilizing the Wireshark application got 4 IP addresses for the attacker.

VII. CONCLUSION

The Internet is one of the fundamental necessities of society, yet it very well may be effectively attacked. Generally speaking, through this venture, we planned to completely appear and portray how hazardous a focused on DoS/DDoS attack can be in the present mechanical world through running the open-source DoS UDP Packet Flood and reenacting a DoS attack. Log file information with p.cap expansion can be examined by network forensic examination utilizing the Wireshark application. We presume that the current IoT systems must fuse the forensic arrangements inside its design to guarantee a sheltered and secure condition. In this paper, we had done the network forensics in IoT forensics investigation for detecting DoS/DDoS flooding attacks on the Internet of Things (IoT) devices.

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Software Defined Network

Analysis of Availability Model Based on Software Aging in SDN Controllers with Rejuvenation

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Abstract

Deficiency of flexibility and programmability of legacy network architecture has been the concern of many networking admirers for some years. Software defined networking (SDN) is a new emerging concept to logically centralize the network control plane in order to automate the configuration of individual network elements. However, the failures of SDN controller are every large impact on the network performance and availability. There are different failure modes in SDN controller outages such as hardware and software. Unplanned downtime is mostly caused by software failure due to software aging rather than hardware failure. The aging related faults have a huge effect on the availability for software components, SDN controllers. For that reason, the work presented in this paper offers the availability model for software aging of SDN controllers by applying software rejuvenation. A stochastic reward net (SRN) is proposed to evaluate the availability assessment of a cluster of SDN controllers. And then how software rejuvenation can improve the performance of SDN controllers is studied. To evaluate the availability of proposed model, mathematical analysis is performed.

Keywords: *Software Defined Networking, Availability, Software rejuvenation, SDN controller*

I. INTRODUCTION

Software defined networking brought a revolution in computer networks that is offered to manage the entire network to be more flexible and programmable [1]. The central approach for SDN is built on a single controller which can be managing of all the node in the infrastructure. However, it can be a single point of failure. Distribute SDN controllers face with all the above issue [2]. The availability of SDN controller is more important issue for the overall system availability and network performance. However, SDN controllers', software component replication mechanism is not a good solution to improve the availability, since the main issue of the failure is often shared among the replicas, for example a bug in a software code [3]. It can suffer unplanned downtime due to computer failure, network failure and software failure and so on. One of the causes of unplanned software outages are the software aging phenomenon due to the degradation of software. There are aging related faults, Mandelbugs, which imitate the gradual degradation of the system performance, because memory leaks, data corruption and accumulation of numeric errors, etc [4]. The most effective solution to handle software failure due to software aging is software rejuvenation. Since the preventative action can be done at optimal time interval, it reduces the cost of system downtime and gets higher availability compared to reactive recovery from failure.

Moreover, a cluster of SDN controllers' architecture is focused in order to improve high availability purposes for the network. ONOS, Open Network Operating System, is a newly released open-source SDN controller which is used to consider in this work.

And then availability model of SDN controller using stochastic reward net is illustrated in order to evaluate the availability of cluster SDN controllers. In order to solve the software failure, preventative maintenance (such as software rejuvenation) is applied for enhancing the performance of proposed model. To show the performance of the proposed method, analytic analysis is presented. To evaluate the models throughout both analytic analysis and then SHARPE tool [5] are considered.

The organization of this paper is as follows. Section 2 provides an overview of the related work. The proposed research approach for how to solve software aging of a cluster of SDN controllers is

presented in section 3. The proposed model for enhancing system availability follows in section 4. Finally, the conclusion is described in section 5.

II. Related Work

In this section, selected publications are reviewed which are related to this work.

The researchers [6] presented that SDN is developed to afford more effective configuration enhance performance and more flexibility for huge network designs

The researchers focused on the state-of-art ONOS controller, designed to scale to large networks, based on a cluster of self-coordinating controllers and concentrate on the inter-controller control traffic in [7]. Vizarreta et.al [4] presented Failure Dynamics in SDN controllers' model and evaluates the impact that different controller failure modes have on its availability. In case study, they showed how the proposed model can be used to estimate the controller steady state availability, quantify the impact of different failure modes on controller outages, as well as the effects of software aging, and impact of software reliability growth on the transient behavior.

The authors [1] presented Software Defined Networking (SDN) brought an unprecedented flexibility and programmability into computer networks. And then they evaluated the benefits by enhancing the ONOS SDN-IP application with an adaptive Robust Traffic Engineering Algorithm.

The authors [8] describe that many software failures are those due to software aging phenomena. So, they proposed a new framework for predicting in real time the time-until-crash of web applications which suffer from software aging using machine learning techniques.

Ros et. al [9] have shown that in order to achieve the availability of five nines, the forwarding devices are required to connect at least two controllers, for all wide area network include in their study. Studies [9] and [4] separate between permanent and transient hardware and software failures.

In the following section, an availability model for software aging of SDN controllers is proposed using stochastic reward nets model and analyze the availability for cluster of SDN controllers in case of software aging failure.

III. Proposed research approach for Software aging in SDN controllers

With the aim of making networks not only more programmable but also easier to manage, SDN based network architecture has been presented. This system consists of a cluster of ONOS controllers and then the exchange of routing traffic among the controllers as shown in Figure 1. In this SDN based network scenario, the configuration, ONOS 1.2 [10, 11] controllers are considered. It is based on all in one SDN hub VM. To imitate the network controlled OpenFlow protocols is used.

Each network device can connect to multiple ONOS controllers, but each domain has its primary controller with full control for forwarding tables and so on. Each controller interacts with all the other controllers and then they send and accept keep-alive messages among controllers for monitoring in a cluster member. If one controller fails, the other controller in a cluster can take the operation from the affected controller. Controller failure frustrates the network ability to serve new requests coming from the network application. Moreover, after node or link failure, it needs to be routing or rerouting for the physical network. There are different failure modes in controller occur with different occurrence. In this work, node failure cause of software aging failure is considered on SDN controllers. Therefore, in order to detect and solve the software aging in case of memory exhaustion in SDN controllers, management server which includes aging detector and rejuvenation manager is considered. Accordingly, in this model two kinds of preventive maintenance (software rejuvenation methodology) have been considered and evaluated.

Taking the rejuvenation action, there is a downtime. For this reason, the rejuvenation action can be done at optimal times in order to reduce the cost of system downtime. The main process of aging detector and how rejuvenation policy used to solve the aging problem are discussed in subsection A and B.

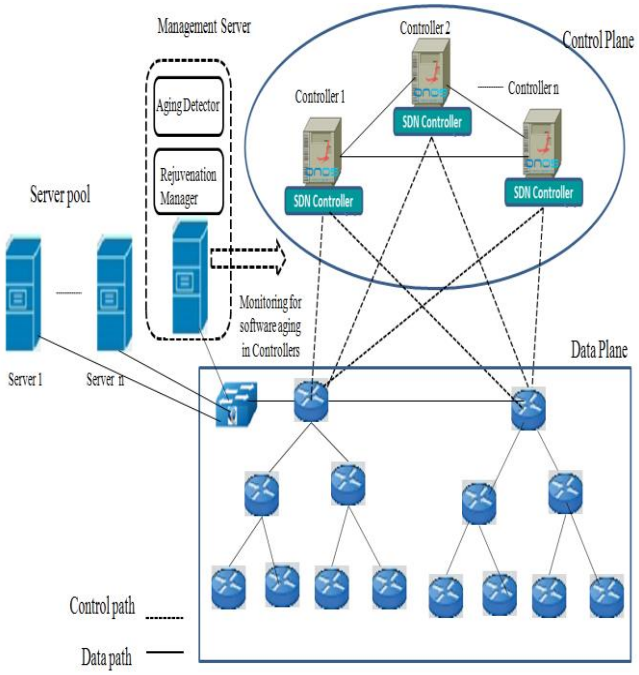

Figure 1. System Architecture

A. Aging Detector

In the management server, the Aging Detector module make the detection of software aging. The responsibility of the aging detector, it can detect the resource exhaustion for memory due to the amount of traffic exchange among a cluster of controllers and updating their states.

SDN controller's failures related to the software aging are detected from both configuration and network. In this work, Mandelbugs in SDN controller is considered. So, the Aging Detector is collecting the resource status Memory, and number of threads or number of connections among the SDN controllers. Because of these parameters can estimate the service time to crash due to software aging. When the aging detector detects resource exhaustion, the proposed rejuvenation policies are applied. Some potential failure happens in SDN controller, the rejuvenation manager that is software component in management server triggers the partial and full rejuvenation according to the failure and estimation of time to crash that detected form Aging Detector.

The work of aging detector and rejuvenation manager is shown in the sequence diagram as figure 2.

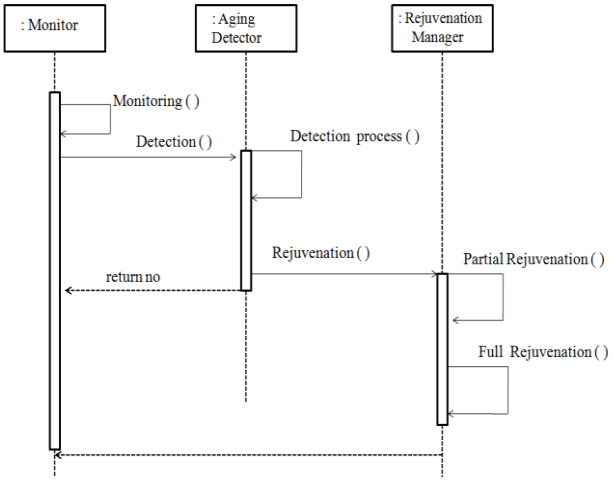

Figure 2. Sequence Diagram for Aging Detector and Rejuvenation Manager

B. Rejuvenation Policy

When Aging Detector detects software aging happens in each SDN controller, the rejuvenation manager will activate the rejuvenation action. The two kinds of rejuvenation policies will be performed on that system. It is based on the condition of the unstable state a minimal maintenance (a partial system clean and restart) or a major maintenance with rejuvenation (clean and restart) is applied.

According to the collecting the resource from Aging Detector, it can be detected some condition could not work well. At that time minimum maintenance is conducted. It is called partial restart and can restore its not working or out of order service such as configuration or Database connections and some other resources back to a healthy state. During the minimal maintenance, controller can provide continued services because this maintenance is a partial system cleanup.

Then, Aging Detector estimate the time to crash of controllers that have violated the threshold (Time Limit) defined by the system administrators per each service or per the whole framework, the rejuvenation manager triggers the major maintenance that is full rejuvenation. So, OS and all services on that controller must be stopped and the rejuvenation action restarts the OS to recover all its free memory. This rejuvenation is also called full restart. When one of the controllers in cluster needs to do major maintenance with rejuvenation, another SDN controller in cluster is to take the responsible for that rejuvenation triggered controller.

When software rejuvenation is performed the system stops serving resulting in a downtime that can be affected the operational costs. But due to its proactive feature, the downtime due to rejuvenation is less than downtime due to an unscheduled software failure. Furthermore, partial rejuvenation can be less downtime cost than full rejuvenation. Therefore, two kinds of rejuvenation (preventative maintenance) are considered in order to reduce system downtime in case of software aging.

IV. STICHASTIC REWARD NETS (SRN) MODEL FOR PROPOSED SYSTEM

A stochastic reward nets (SRN) model with two kinds of rejuvenation mechanism for a cluster of SDN controllers is described in Figure 3. It has n tokens which represented n SDN controllers in cluster. In initial condition, all controllers are working well state, indicated by a token in place P_{Up}. When transition T_I fires the controller enters the inspection state and a token moves from P_{Up} to P_I. After inspection is complete (firing the transition T_{up},) no action is taken if the system is found to be in working state. Transition $T_{U,I}$ models the unstable state of the controller. When this transition fires, (i.e., a token reaches place $P_{U,I}$) the controller is operational but in the unstable state. The transition $T_{U,I}$ models the unstable state and under inspection state of the SDN controller. At that time, some service in SDN controller is not working well but it is still running. So, it needs to a partial system clean and restart for SDN controller. After minimal maintenance is complete (firing the transition T_m), the controller enters the operational state.

As the time progress, each controller can eventually transit to software aging state in place P_{FP} detected from the aging detector as memory leaks which reproduces the gradual degradation of the system performance through the transition T_{Aging}. If the one of the SDN controller is about to be major maintenance with rejuvenated, the traffic control and the routing decision of the packet on affected controller is switched to another controller in cluster and then the effected controller will be started for the new requests and sessions before rejuvenation through the transition T_{sw}. It can return to the original SDN after the accomplishment of the rejuvenation such as system clean and restart.

Consequently, the rejuvenation interval is defined through clock with guard function ginterval. The tokens are in the place P_{clock} and P_{FP}. If there is a token in $P_{trigger}$ through guard gtrig, and there are controllers to be rejuvenated. In that state token is placed in P_{FP}, immediate transition t_{rej} is enable through the function grej as shown in Table 2. After the controller has been rejuvenated, it will be in the working state through transition T_{H_rej}. And then, immediate transition is enabled by using greset and a token is switched to P_{clock}.

If the software aging failure cannot be detected, the controller can fail because of software aging and the controller is not working in the place of P_{Aging}. At that time, there is no active controller in a cluster, the system may outage and then the system goes to Down state in the place of P_{Down} through the transition T_{Hfail}. From a whole system down, the system can be restoration with the transition T_{Repair}. If one SDN controller is used in system architecture, the downtime can be more because it does not switch to routing decision and some other services to another active controller. Otherwise, there is an active controller in a cluster; the controller can be replaced through the transition $T_{Replace}$. After repairing the fail controller through T_{Repair}, the controller can join in the cluster.

The Markov reward model which is mapped from proposed SRN constructs a marking dependency and enabling functions (or guards) facilitate the construction of model in order to solve the steady state availability for proposed model. Then, the SRN model has been evaluated through numerical derivation that can be shown how rejuvenation mechanism can reduce downtime and the architecture for a cluster of SDN controllers can improve the system availability with the mapping of reachability graph in sub section A

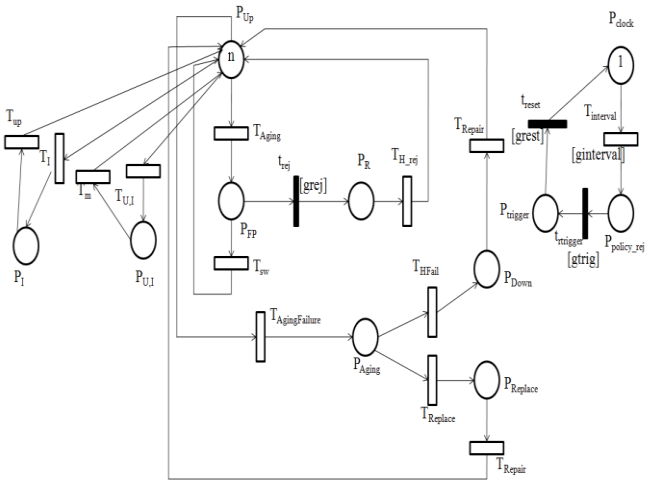

Figure 3. Stochastic Reward Nets Model for Software Aging in SDN controllers

TABLE I. DESCRIPTION FOR PLACES OF PROPOSED MODEL

Places	Description
P_{Up}	A cluster of SDN controllers Up state
P_I	Inspection state in SDN controllers
$P_{U,I}$	SDN Controllers unstable state under inspection
P_{FP}	Software Aging state in SDN controller
P_R	Proactive Software Rejuvenation state in SDN controller
P_{Aging}	SDN controller Software aging failure state
$P_{Replace}$	SDN controller replaced state
P_{Down}	Failure state of all SDN controller in cluster
P_{clock}	Rejuvenation interval state
P_{policy_rej}	Rejuvenation policy state
$P_{trigger}$	Rejuvenation state

A. Reachability Analysis

In this section, the reachability graph is constructed for the mapping of proposed model.

Let 7 tuples (π_{Up}, $\pi_{I\,i}$, $\pi_{U,Ii}$, $\pi_{R\,i}$, π_{Aging}, π_{Down}, and $\pi_{Replace\,i}$) denote the marking with π_x =1, if a token is presented in place $\pi_{x,}$ and zero otherwiseas shown in figure 4.

This figure illustrates with squares representing the markings and arcs representing possible transition between the markings. Let λ_I, $\lambda_{U,I}$, μ_{up}, μ_m, λ_R, μ_{rej}, $\lambda_{sw\text{-}age}$, $\mu_{replace}$, $\lambda_{fail}$and μ_{repair}be the transition rates associated with T_I, $T_{U,I}$,T_{up}, T_m, T_R, T_{H_rej}, $T_{AgingFailure}$, $T_{Replace}$, T_{Hfail}, and T_{repair}respectively. By mapping through actions of this graph with stochastic process, steady-state solution can be achieved.

TABLE II. GUARD FUNCTION FOR PROPOSED MODEL

Name	:	Guard Function
ginterval	:	(#(P_{FP}) = =1)
grej	:	(#($P_{trigger}$) = =1)
greset	:	(#(P_R) = =1)
gtrig	:	(#(P_{policy_rej}) = =1)

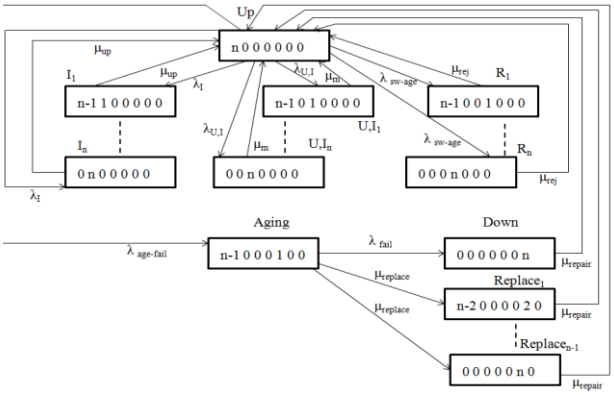

Figure 4. Reachability Graph for the proposed model

The summing the probabilities of all states in proposed system equation is described in Equation 1.

$$\pi_I + \pi_{U,I} + \pi_R + \pi_{Aging} + \pi_{Down} + \pi_{Replace} + \pi_{Up} = 1 \quad (1)$$

The combination of balance equation that obtained from figure 4 with the above the summing of probabilities, the closed form solution can be achieved.

$$\pi_I = \sum_{i=1}^{n} {}^{\lambda_I}/_{\mu_{up}}\, \pi_{Up} \quad (2)$$

$$\pi_{U,I} = \sum_{i=1}^{n} {}^{\lambda_{U,I}}/_{\mu_m}\, \pi_{Up} \quad (3)$$

$$\pi_R = \sum_{i=1}^{n} {}^{\lambda_{sw-age}}/_{\mu_{rej}}\, \pi_{Up} \quad (4)$$

$$\pi_{Aging} = A\pi_{Up} \quad (5)$$

$$\pi_{Down} = {}^{\lambda_{fail}}/_{\mu_{repair}}\, A\pi_{Up} \quad (6)$$

$$\pi_{Replace} = \sum_{i=1}^{n-1} {}^{\mu_{replace}}/_{\mu_{repair}}\, A\pi_{Up} \quad (7)$$

$$\pi_{Up} = \left\{ 1 + \left[\begin{array}{l} \sum_{i=1}^{n} \lambda_I / \mu_{up} + \\ \sum_{i=1}^{n} \lambda_{U,I} / \mu_m + \\ \sum_{i=1}^{n} \lambda_{sw-age} / \mu_{rej} + \\ A + \\ \lambda_{fail} / \mu_{repair} A + \\ \sum_{i=1}^{n-1} \mu_{replace} / \mu_{repair} A \end{array} \Big/ \begin{array}{l} \sum_{i=1}^{n} \lambda_I / \mu_{up} \\ + \sum_{i=1}^{n} \lambda_{U,I} / \mu_m \\ + \lambda_{sw-age} \\ + \lambda_{age\_fail} \end{array} \right] \right\}^{-1} \quad (8)$$

Where A= $\lambda_{age\_fail} / (\lambda_{fail} + \mu_{replace})$

TABLE III. THE MEANING OF THE PROBABILITY'S ARE AS FOLLOWS.

π_{Up}	:	The probability of SDN controllers are working state
π_{I_i}	:	The probability of SDN controller is under inspection state
π_{U,I_i}	:	The probability of SDN controller is not working well state
π_{R_i}	:	The probability of SDN controller is Rejuvenation state
π_{Aging}	:	The probability of SDN controller is software aging failure state
π_{Down}	:	The probability of SDN controllers are Down state
$\pi_{Replace_i}$	:	The probability of SDN controller is replace state

B. Analysis of Availability and Downtime

In this subsection, according to the numerical derivation, availability and downtime are evaluated. Availability is a probability of SDN concept-based network infrastructure which provides the services in each instant time through reachability graph. In this model, SDN controller services have been focused. Based on the proposed model, the whole system may be totally down when all controllers failed because of software aging failure and there is no hardware extra for replace SDN based network. The system is not available in Down state, there is a token in the place (P_{Down}). Downtime is the expected total downtime of a cluster of SDN controllers-based network in an interval of T (24*30*12 days). The system availability in the steady-state and Downtime are defined as follows:

$$Availability = 1\text{-}Unavailability \quad (9)$$

$$Availability = 1\text{-}\pi_{Down} \quad (10)$$

$$Downtime\ (T) = T^{*}\ \pi_{Down} \quad (11)$$

The applicability of the proposed model and solution methodology through numerical examples are illustrated using the transition firing rate. A good estimate value for a range of the model is assumed because the exact transition firing rates are not well known normally. So, experiments are performed using the following failure profile in literature [9, 4, and 12] mentioned in Table 4.

TABLE IV. PARAMETERS VALUES AND DESCRIPTION

Transition	Description	Values
T_I, $T_{U,I}$	Inspection of Aging Probably	1time/a day
T_{up}	After Inspection without maintenance Rate	0.3
T_m	After Inspection with maintenance Rate	0.6
T_{Aging}	Software Aging Rate	1 day
$T_{AgingFailure}$	Failure Rate that effect of aging	7 days
T_{sw}	Switch over rate	5 secs
T_{HFail}	Hardware/Physical host failure rate	6 months
T_{Repair}	SDN Controller Physically Repair Time	24 hrs
$T_{Replace}$	Replacement Time for Extra SDN Controller	2 hrs
$T_{interval}$	Rejuvenation interval	2times/a month
T_{H_rej}	Rejuvenation rate	2 mins
T	Unit Time Interval	24*30*12 days

The influence of number of SDN controller along with different failure rate caused of software aging on availability is shown in Figure 5. It can be seen the result that the need of at least two or three SDN controllers for achieving "five-nines" availability. So, the availability is dependent on the number of SDN controllers. According to the analysis,

the higher mean time to software aging failure of SDN controller, the higher availability can be obtained.

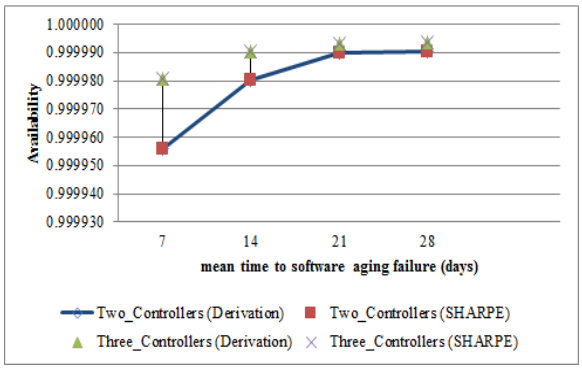

Figure 5. Availability vs different software aging failure time for multiple SDN controllers

The Figure 6 shows the differences in downtime with different number of SDN controller in cluster through different controller software aging failure time. The more controllers, the lower downtime can be obtained. As a result, the downtime decreases up to 5 minutes for three SDN controllers. According to the analysis, the consideration of a cluster of multiple controllers can be affected on system downtime. Therefore, cluster of SDN controller's architecture are applied in this work.

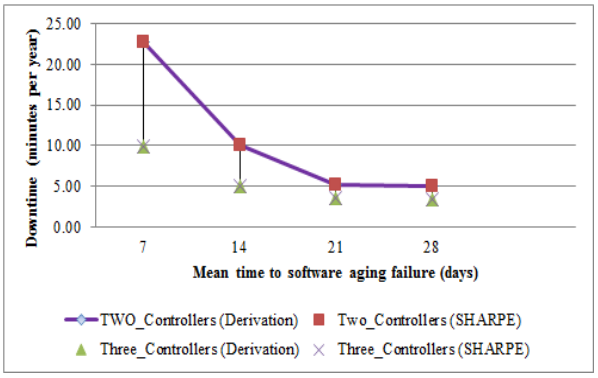

Figure 6. Downtime vs different software aging failure time for multiple SDN controllers

In our proposed research approach, two kinds of proactive rejuvenation are used for counteracting the software aging affected on SDN controller. With the purpose of it, the inspection for probably of software aging in SDN controller is very important because it can be noticed that the rejuvenation action applies or not. So, the influence of mean time between inspections (MTBI) along with different controller of software aging failure is evaluated through the proposed model. The analysis result of availability using above mentioned system-parameters is shown in Figure 7. There are several different values of time to carry out the inspection. The MTBI are assumed 6 hours and 12 hours. The lower MTBI, the higher availability can be reached.

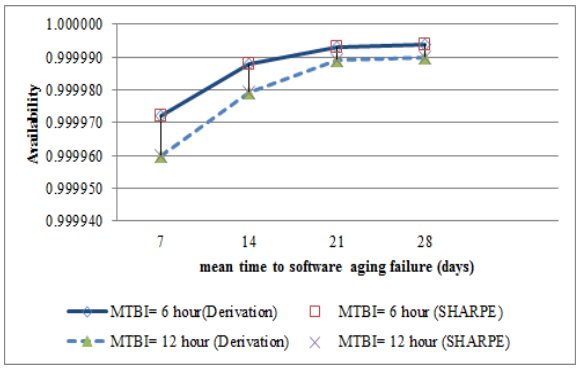

Figure 7. Availability vs different software aging failure time with different Inspection time for aging

The Figure 8 shows the differences in downtime with different software aging failure using various MTBI. From the result, the lower MTBI for SDN controller, the lower downtime can be obtained. For the best and most expensive of all SLA services, the downtime will be up to five minutes per year and application uptime would also be about 99.999% of the time. According to the downtime analysis, the inspection time for probably of software aging is important in SDN controller.

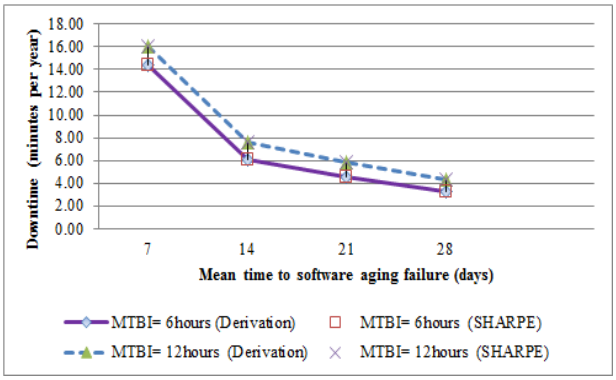

Figure 8. Downtime vs different software aging failure time with different Inspection time for aging

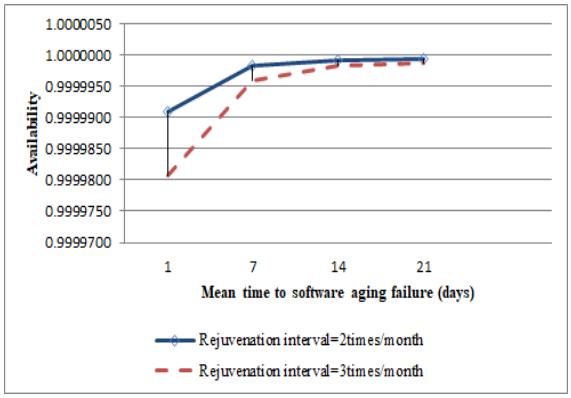

Figure 9. Availability vs different software aging failure time with different Rejuvenation interval

The Figure 9 shows the differences in availability with different software aging failure between once a week starting one day using various rejuvenation interval. From the result, if the rejuvenation did many times, the lower availability will be obtained. According to the analysis results, the rejuvenation interval should be optimal time interval and it is probably important for different software aging failure in SDN controller.

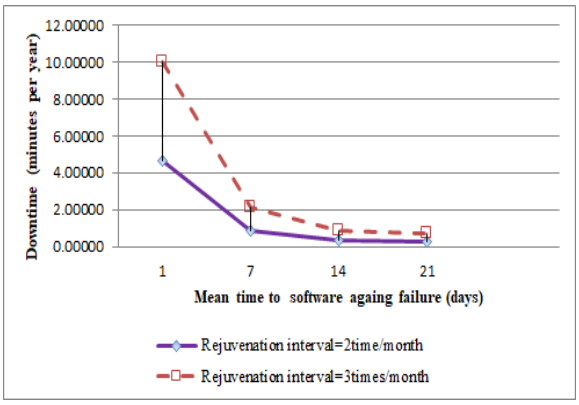

Figure 10. Downtime vs different software aging failure time with different Rejuvenation interval

The differences in downtime with different software aging failure using various rejuvenation interval is shown in figure 10. According to the analysis results, the rejuvenation interval should be optimum interval because the more rejuvenation, the lower downtime can be achieved especially software aging failure in a week.

According to the analysis results shown in figures, the failure rate caused of aging has also an impact on the numerical analysis of availability and downtime of the proposed model. However, aging failure rate is not accurate parameters, because it depends on the many factors such as controller utilization rate such as CPU, memory and so on. In our studies, the cause of software aging is based on memory exhaustion because SDN controller responsible that the control plan of forwarding networking devices is extracted and moved to the entity. Therefore, the varying aging failure rates are considered to evaluate the proposed model. The software aging caused by other factors like CPU utilization can be considered with different software rejuvenation interval in future.

According to the figures, it is shown that the derivation results and SHARPE [13] simulation results are the almost same. Therefore, it can be proved that the numerical derivation results for proposed model are acceptable.

V. CONCLUSION

This research work is intended for SDN based network infrastructure meant for improving the network performance. The objective is specially associated with how to solve software aging failure in SDN controllers which can be impact on network performance. Based on the numerical analysis using failure profile, a stochastic reward nets model for software aging of SDN controllers is evaluated in available and downtime. As the obtained results shown in figures, it can be said this approach get the high availability and lower downtime (uptime about 99.999% of the time) when the number of SDN controllers increased.

Two kinds of rejuvenations can also enhance the availability of SDN based network with optimal rejuvenation interval that can reduce the downtime of the system as shown in results.

Although the (n) numbers of controllers in a cluster are deployed in the proposed model, evaluation result show that multiple SDN controllers, at least two or three controllers offer the high availability of the services and in order to achieve minimizes downtime than traditional network. For future work, how software aging failures influences the impact on SDN based environment in other SDN controller instead of ONOS.

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Flow Collision Avoiding in Software Defined Networking

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Abstract

With the high volume of traffic in recent network, traffic between the switches has exchanged rapidly among the servers that may conduct the link congestion. In a traditional network, routing protocols are used the shortest path algorithm and congestion will occur in the network links. Equal-Cost Multi-Path (ECMP) algorithm cannot fairly distribute the bandwidth of the links between the flows. These flows suffer long queuing delays and degrades throughput. Flow Collision Avoidance algorithm (FCAA) is an efficient method to alleviate the network congestion by rerouting the flows to guarantee the performance of the network. Congestion occurs when flow demand exceeds link capacity and it leads to the degradation of QoS. The algorithm evaluates the existing flow's demand based on port statistics to handle the collision of the large flow in the network avoiding the congestion. The evaluations indicate that SDN-based FCAA algorithm enforces the required end-to-end QoS for each traffic flow over ECMP.

Keywords: *ECMP; FCAA; SDN; QoS*

I. INTRODUCTION

Software Defined Networking can enable an effective programmable solution to allow flexible network for the current enterprise network. SDN is an architecture that reduces the network operation costs and optimizes the network resource usage. In SDN, network monitoring can be accomplished using OpenFlow statistics and the controller can keep track of available bandwidth on each link. SDN increases higher rates of innovation and it decreases the obstruction for a new technology in large scale network. The SDN represents a network framework with programmatic ways to control forwarding function.

ECMP is a networking strategy for load balancing to allocate flow to get an available path. ECMP is per-flow, hash based traffic distribution. It routes the flows on to the paths over static hashing. It is effective for small flows but it is not operative for large flows due to static hash collision. ECMP is a forwarding mechanism to forward packets along the multiple paths of equal cost with the goal to get distributed link load sharing. These static flows do not consider for resulting collisions degrading overall utilization for existing network. To guarantee the desired overall performance a service, network operators need to offer accommodations for QoS applications.

The SDN Controller gets the commands using OpenFlow statistics and interconnects with the applications including port statistics from the network devices. OpenFlow allows network traffic control from the controller and it also offers many concepts like traffic engineering. OpenFlow is a protocol and OpenFlow switches connect with other switches with OpenFlow ports. OpenFlow messages forward the traffic between OpenFlow enabled switches through the network. Each switch maintains a flow table that contains forwarding information. The controller and switches communicate via OpenFlow messages. OpenFlow controller can arrange data flows by replying the OpenFlow messages from switches. By using the OpenFlow protocol, an OpenFlow controller can insert flow entries into OpenFlow allows network traffic control from the controller and it also offers many concepts like traffic engineering. Based on the traffic statistics along with the network bandwidth information, a controller can construct the network topology. The controller provides network control decision and network devices to forward packets through the traffic statistics.

To get the alternate path for the absence of failures, the existing forwarding protocols are improved. Congestion control is a key factor in ensuring network stability and robustness. In this paper, this paper compares FCAA against ECMP to handle the collision of the large flow in the network avoiding congestion. FCAA can also minimize the network congestion by rerouting the flows over the alternative paths. FCAA algorithm compute alternate light loaded path by forwarding the flow to avoid collision for the large flows.

The rest of the paper is designed as follows:

Related work is explained in Section II. Then this paper discusses flow congestion avoidance scheme in Section III. The evaluation results are discussed in Section IV. In Section V, this paper concludes and future work is provided.

II. RELATED WORK

The proposed method is based on Traffic Engineering. In this section, this paper will discuss the literature works.

In [1], Ian F. Akyildiz, et al. provides an overview of traffic engineering mechanisms to manage data flow efficiently at both the control plane and the data plane in SDN architectures. They also discuss classical TE mechanisms developed for ATM, IP and MPLS networks, and then survey in detail the state-of-the-art in TE for SDN from both academia and industry perspectives.

An efficient method based on SDN discussed for reduction of congestion in data-center networks for rerouting of selected network flows in the switches with the congested links [2].

The authors proposed a method to reduce congestion in SDN data center networks. The controller finds the costs of each link and rerouted the flows through alternate paths with minimum load. The authors in [3] do not consider the overhead for the controller when the controller computes the link utilization and link load of each path.

The authors presented that link utilization is calculated in the SDN controller and recalculated rerouting algorithm is applied to switches which would be configured by using OpenFlow configuration protocol [4].

In [5], the authors proposed an efficient reallocating method for SDN based on genetic algorithm. This paper is compared with ECMP in terms of throughput, packet loss and data transfer to minimize network congestion.

The authors in [6], the problem proposed to obtain improved allocation of resources using the Genetic Algorithm but it is not able to realize the parallel optimization.

The paper in [7] utilizes the hierarchical fat-tree network to efficiently schedule flows based on traffic statistics that deliver multiple alternative paths among hosts and to handle the network congestion via OpenFlow protocol.

The authors in Mahout [8] proposed to monitor and detect the large flow on the terminal host through a shim layer in the operating system rather than directly detecting the switches in the network.

III. Flow Congestion Avoidance in SDN

The concept of SDN enables an efficient network configuration by separating the control logic from the infrastructure network supporting centralization. The control plane controls the behavior of the network by installing forwarding rules to the infrastructure layer. Finding the optimal paths for traffic demands is a problem to define routes dynamically in traffic engineering (TE). Traffic engineering optimizes the performance of the network and avoids the congestion on any one path. It guarantees bandwidth guarantee in the network links. It has the capability to distribute QoS issue traffic when the network congestion occurs.

SDN enables the network control to convert directly programmable and centralized management and the infrastructure layer abstracts the network applications and network services. The SDN makes simpler the network management to reduce operating costs and support innovation. The SDN controller monitors traffic statistics to install forwarding rules into the switches as shown in Figure 1. Traffic measurement in SDN is a vital task to gather statistical data about network flows from the switches in real-time.

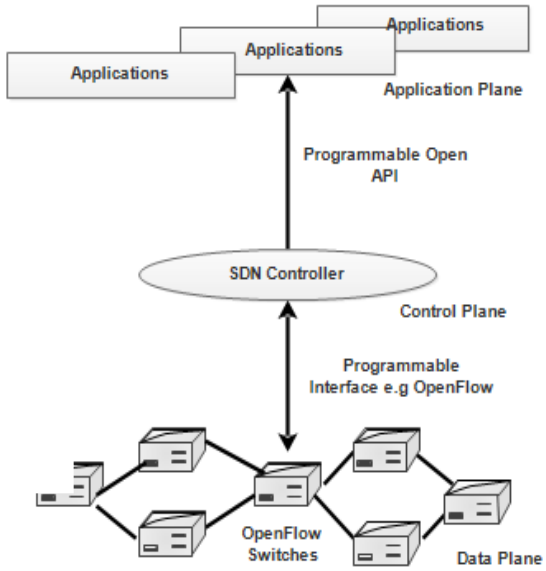

Figure 1. SDN Architecture

The SDN Controller gets the information using OpenFlow statistics. OpenFlow provides access to the infrastructure layer over the network and it forwards the packets between the switches. The controller propagates the forwarding rules to all switches in the infrastructure layer.

Traditional enterprise networks use Equal Cost Multipath (ECMP). It is used to allocate a path for each new flow that depends on hashing. It also performs the static traffic splitting and it can collide on the switches between the large flows. So it can degrade the efficient resources utilization. ECMP cannot handle to select the path dynamically. This paper compares FCAA against ECMP to handle the collision of the large flow in the network avoiding congestion. FCAA considers rerouting flows when the network congestion occurs over the alternative links. It provides the efficient routing scheme to select a new path for the large flow with the minimum congestion when link congestion occurs.

The main contribution of this paper is to compute alternate light loaded path by forwarding the flow to avoid collision for the large flows and the FCAA algorithm that is outperformed the conventional ECMP in fat-tree network.

The controller sends Statistics Request message to the switches to achieve the required port statistics. The switches reply to the controller with Reply message. Then the

controller provides the decision for the new flow to a virtual network composed by queues of appropriate links which meets its performance requirements. The controller computes the light loaded path for the flow and assigns this flow into the switch. When the controller occurs flow entries from switches on the heavy-loaded path, it will shift the flow entries into switches over the light-loaded path.

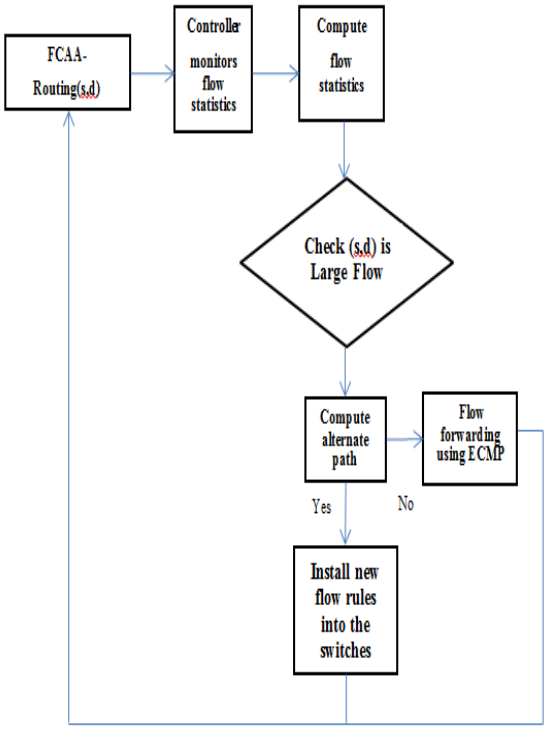

Figure 2. Algorithm Flowchart of FCAA

In the FCAA algorithm, the controller configures the estimated threshold value (10% of link capacity) to switches for classifying the large flow. When the large flow is detected, the proposed system will use the flow collision avoidance algorithm FCAA to deal with the corresponding large flow. The algorithm routes the flows along the lightly loaded path.

The method is to find the light-loaded path for a flow described in the Algorithm as shown in Figure 2. This system uses this information to build up the network G(V, E) where the vertex V corresponds to the switches and the edge E corresponds to the links.

IV. EXPERIMENTAL SETUP

The evaluation was presented to demonstrate the algorithm based on the fat-tree network topology as shown in Figure 3. The VM image has a 64-bits Ubuntu 16.04 installed as the guest OS. The system consider fat-tree of k=4. A fat-tree network has three layers: core, aggregation and edge. Nowadays, the fat-tree network is one of the most widely used topologies. It also consists of two types of elements: core and pods. The Fat-Tree topology consists of 20 switches and 16 hosts. Each pod has k/2 aggregation switches and number of k-port switches. In Ports Statistics Monitoring, the system sets the monitoring period as 5 seconds. And it sets the maximum capacity as 100 Mbps. The metric used to compare the ECMP is the throughput and flow completion time. The Mininet emulator is used to create the virtual networks to model the fat-tree network topology.

The SDN controller in the system is a free OpenFlow controller that runs with the Mininet in the Ubuntu. The algorithm can be applied to the fat-tree network topology as an example. The system uses iperf tool to test the performance. The system runs iperf servers on hosts H1, H2, while two iperf clients are started on each of H9, H10. In the FCAA algorithm, the controller configures the estimated threshold value (10% of link capacity) to switches for classifying the large flow. Lastly, when the measurements of a flow exceed threshold value, the controller determines that the flow is a large flow and it will forward some flows to the light loaded path based on port statistics. In the measurement, this paper has been tested with default window size is 85.3 kB for both tests.

The performance of FCAA is evaluated in terms of throughput. Throughput is calculated using the formula.

$$Throughput = \frac{Amount\ of\ data\ transferred}{Time\ taken\ to\ transfer\ data} \qquad (1)$$

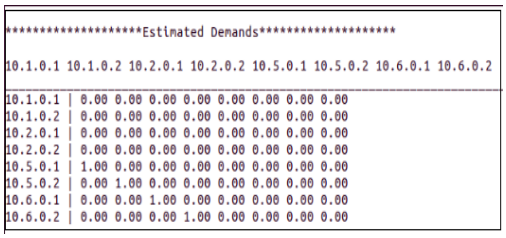

Figure 3. Demand estimation between flows

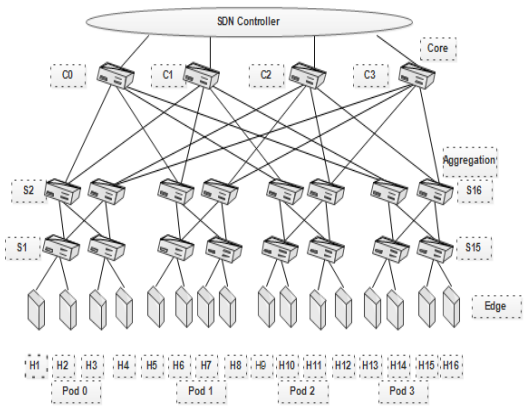

Figure 4. The topology of Fat-Tree network in SDN with k=4

TABLE 1. SIMULATION PARAMETER SETTING

Parameters	Values
SDN Protocol	OpenFlow 1.3
Controller	OpenFlow Controller
Software	Mininet 2.3
Link bandwidth	100 Mbps
Threshold	10 Mbps
Window Size	1 MB

In the testing, flow 1, 2 and 3 from source host H1 and H2 will take different routes to their destination hosts. A host sends packets to another host in the network. The FCAA algorithm is performed by generating the different numbers of flows as shown in Figure. 3. The parameter settings used in the testbed is shown in Table I. FCAA achieves better performance than ECMP. ECMP increases significantly due to congestion. FCAA outperforms ECMP. The system uses Iperf to generate traffic flows among switches in the different pods. Iperf is command line-only tool for the active measurement of the maximum network throughput for IP networks. The system tests the fat-tree network topology using Mininet network emulator as shown in Figure 4.

For example, assuming that flow f from h1 to h9 uses the primary path (S9 → S10 → C1 → S2 → S3), the alternate paths used to keep the link on the main path of flow obtained using FCAA. The algorithm calculates the light loaded path for each congested large flow. Therefore, the congested flow will be rerouted through (S9 → S12 → C2 → S4 → S3).

In this testbed, the throughput between flows from H1 to H15 was measured. In the depicted figure 5, 6, 7, the flow between H1 and H9 are steering the flows in the Flow-1. Then, the two hosts, H2 and H10 are forwarding the flows is shown in Flow-2. The final Flow-3 represents the flow between H3 and H11. The comparison of the experiment results was drawn by using FCAA and ECMP for the throughput of flow1 in Figure 5. According to the testing, the performance of FCAA flows increase quickly from 57.7 to 94.8. ECMP flows increase slightly to 87.4. The results produced by ECMP are receiving poorer due to congestion.

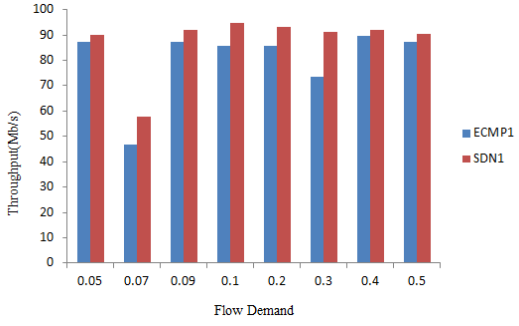

Figure 5. Throught result for Flow1 between Demand estimation

The comparison of the experiment results was depicted by using FCAA and ECMP for the throughput of flow2 in Figure 6. According to the testing, the performance of FCAA flows increase quickly from 82.2 to 95.6. ECMP flows increase slightly to 89.6. FCAA flows have the potential to improve long flow throughput at flow demand 0.1.

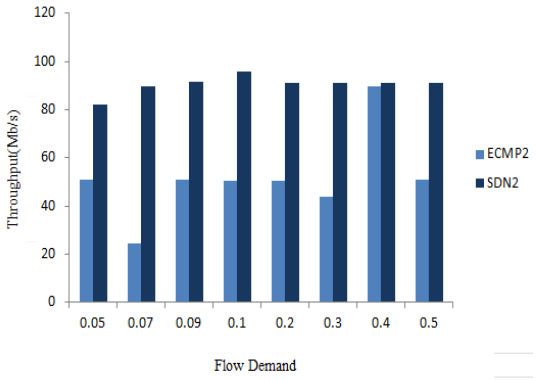

Figure 6. Throught result for Flow2 between Demand estimation

In figure 7, the comparison of the experiment results was presented by using FCAA and ECMP for the throughput of flow3. According to the testing, the performance of FCAA flows increase quickly from 32.7 to 67.7. The results produced by ECMP are receiving lesser due to collision. FCAA flows also can still balance the loads of flows more efficiently and have higher throughput at flow demand 0.1.

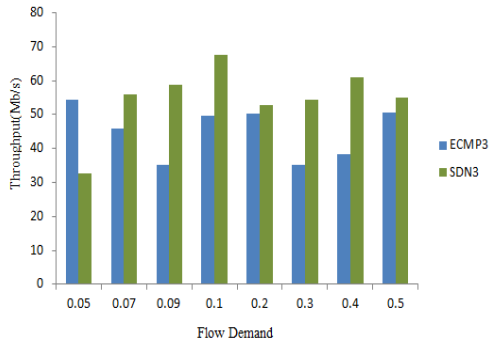

Figure 7. Throught result for Flow3 between Demand estimation

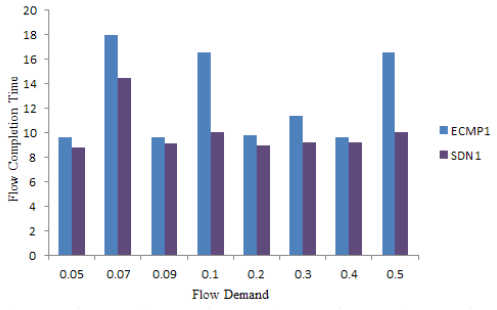

Figure 8. Flow Completion Time between Flows

The test result of the flow completion time FCT for FCAA is shown in Figure 8. The FCT for ECMP increases

from 9.6 to 16.4. In FCAA, FCT increases slightly from 8.8 to 14.5. For flow completion time, FCAA achieves better performance than ECMP.

V. CONCLUSION

In this paper, the traffic-based collision avoidance method was implemented that reduces network congestion according to their flow demands and flow statistics based on SDN. It can deal with the problem of large flow collision and the traditional ECMP. For the collision of large flows, the effect of FCAA collision avoidance was also presented to avoid congestion and focused on the problem of ECMP through fat-tree network topology. For the experiment, the Mininet emulator is used to evaluate the algorithm by comparing ECMP. The performance of the FCAA algorithm has the potential to improve higher throughput than the conventional flow hashing-based ECMP. In conclusion, more research is necessary to calculate optimized paths and to develop collision avoidance approaches to find collision free traffic. As for the future, the system will be planned to improve the congestion avoidance of fat-tree network management and to improve the effectiveness of network resource utilization.

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Market Intelligence Analysis on Age Estimation and Gender Classification on Events with deep learning hyperparameters optimization and SDN Controllers

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Abstract

Nowadays, understanding demographic information on social influencer events is important for target customer analysis. Hence monitoring the crowded event requires an intelligent sophisticated technology with human gender classification and classify age group. This paper is using hyper-parameter optimization and SDN controller on age gender classification on events cameras to monitor and classify to cover the whole event. Many cameras will need and lead to weak performance in classification. This system targets for getting better result in multi-IP-cameras managing port under SDN controller. Classification for image processing is used by fast R-CNN with hyper-parameter optimization model data training and achieving result from SDNs based web cameras. With constraints of hardware, this system will rely on group of Sony IP-Cameras with SDN controller environment. And OpenCV2 libraries is used to train hyper-parameter optimization model, SDN controller run on Mininet , Openflow protocol and IMDB Image Datasets and Asia Image Dataset are used to achieve better satisfactory performance.

Keywords: *Gender classification, IP-Camera, SDN, OpenCV, OpenFlow, IMDB image Datasets*

I. INTRODUCTION

Present Days, Business Analytics is main driven to not even business planning but also in Government national planning. Hence, collection of data and data mining are critical parts in demographic data. Video analytics application helps in providing information as people counting, evaluate impact of advertising and determine optimal decision.

The system proposed to detect and compute in human gender classification, tracking and counting people with different age classes in events streaming video. Gender classification and age estimation automated systems have become relevant to many applications, particularly since the rise of social media and platform events [1]. In private event, many cameras will need in monitoring all entrances and exits and input to detection system. There is challenge in accuracy of classification and delay due to traditional network will leads to system performance and reliability. The growing complexity of fast R-CNN architectures causes many problems, an important one is ”overfitting” and this happens when the network trains the dataset, but is not able to recognize slightly different objects outside of the dataset.

This study focuses on improving performance of classification task can be defined as recognition rate, i.e. percentage of correctly classified images. The performance can be influenced not only different datasets, but also changing training parameters and additional parameters as hyper-parameters. To get optimized hyper-parameters, this study used Nelder-Mead method as applied to hyper-parameter tuning problems. This searching algorithm will help to save a lot of tedious effort using some intelligent search strategy. This paper indicated that providing better fast R-CNN architectures than a baseline method and achieving nearly state of the art performance without any user intervention provides better results under certain conditions.

Monitoring events are play role in the growth of business analysis data and placed huge demands on data network. This system also tried to modify video streaming performance by using port management on SDN controller. The system improves the packet delay and loss performance of streaming video over Mininet and Open v-Switch based IP cameras cluster OpenFlow networks.

In this paper has 6 Sections, Section 2 has explained related works. Section 3 includes the brief overall system flow, and explains fast R-CNN, Hyper-

parameter optimization hybrid model and SDN configurations to classify gender detection. Section 4 discusses preparation of datasets and training and Section 5 has detail results and followed by conclusion and future works are under in Section 6.

II. Related Works

Human gender classification and age estimation have been developing many methods and platforms. The major issue of gender prediction is how to extract representation features from face detection and age. As per "Real-time age and gender estimation from face images" by JANG-HEE YOO, SO-HEE PARK (iCMLDE-2017) [2] : Face Detection and pose estimation methods are adopted to acquire frontal face images. One famous method is CNN (Convolution Neutral Network) and the enhancement fast R-CNN proposed by ROSS GIRSHICK (iCCV-2015) [3] some of the drawbacks of R-CNN to build a faster object detection algorithm and similar approach on modified R-CNN algorithm.

To get better performance on fast R-CNN, hyper-parameters can add the pre-processing. CHAIRE HAVAS proposed hyper-parameter optimization on continuous parameters sets for better machine learning. Hence, it can use input as additional hyper-parameter sets in fast RCNN machine learning[4]. It has also compared possible algorithm and methods for better optimization results.

And then SDN (Software define Network) is a logically centralized controller, with a global view of network, which can monitor traffic flows, make forwarding decisions and install efficient rules at runtime. The flexibility control accomplished to provide services like inter-domain network layer multicast. SDN-based multicast frameworks supported to offer content and/or network providers with sufficient control to realize and better result on a video streaming service.

III. Proposed Overall System

As Fig.1, the proposed overall system have two main parts. First, the input streaming videos from many cameras from multiple location and passed over the SDN controller network with optimization streaming flow. The centralized controller of network by separating the control logic to its desired devices such as routers, switches and IP based cameras. The proposed network system can resolve delay video streaming problems with SDN based networks. The controller applications can be used configuring and controlling the network. SDN controller run on POX based controller rely on OpenFlow installed Raspberry Pi clustered and mininet for its local testbed. The proposed system aims to efficient real time video streaming through SDN controller to reroute streaming packets faster and efficient.

The second part is classification of input real time video stream from events (such as music concerts, festival) via SDN based controller to assign deep learning system, which applied Fast RCNN based gender classification for market analysis information.

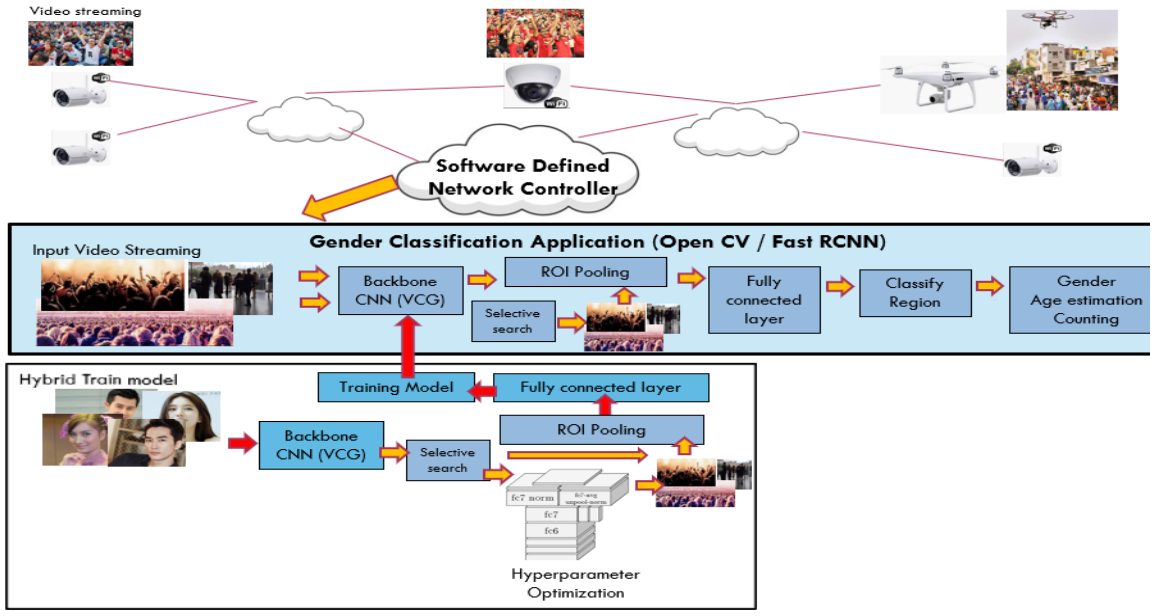

Figure 1. Overall system architecture

And the system used the hybrid system added with hyper-parameter optimization method on

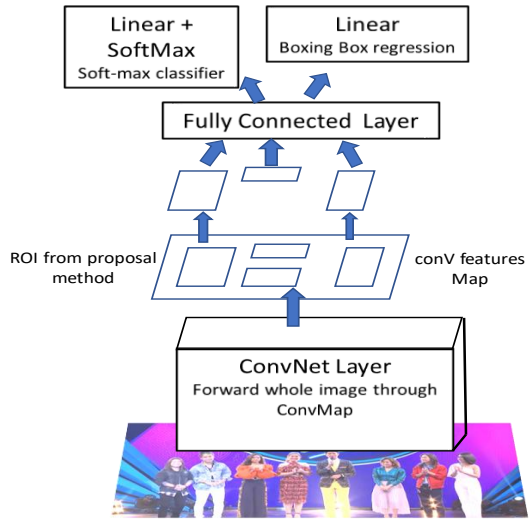

Figure 2. Fast R-CNN architecture

training dataset in preprocessing recursive Fast R-CNN method to get better results as benchmark and accuracy rate is higher with hybrid hyperparameters trained system. The result can show to reduce time consumption and performing to accurate result.

A. Fast R-CNN

As Fig. 2, at the pre-processing stage, the input streaming videos has passed as the input image to the Convolutional Neural Network, which one at a time generates the convolutional feature maps. In view of utilizing these maps, the region extracted due to regional proposal. Then, a RoI pooling layer has applied on all of these regions to reform them as per the input of the ConvNet into a fixed size. Therefore, each region can feed to a fully connected network. On the softmax layer, it has applied on top of the fully connected network and the linear regression layer has applied parallel to output bounding box organized for pre-trained predicted classes.

B. Hyper-parameter Optimization Hybrid Model

In deep learning, a hyperparameter optimization model can be formulated as a stochastic black box optimization model to minimize a noisy black box objective function *f*(*x*)[5].

$$\min_{x \in \mathbb{R}} \ f(x)$$

This system applied Nelder-Mead optimization methods to adjust fewer hyper-parameters. It is developed with fewer computing resources. The

Initialization: Choose an initial simplex of vertices $\mathbf{Y}_0 = \{\mathbf{y}_0^0, \mathbf{y}_0^1, \ldots, \mathbf{y}_0^n\}$. Evaluate f at the points in $\mathbf{Y}_0$.
Choose constants:

$$0 < \gamma^s < 1, \quad -1 < \delta^{ic} < 0 < \delta^{oc} < \delta^r < \delta^e.$$

for $k = 0, 1, \ldots$ **do**
  Set $\mathbf{Y} = \mathbf{Y}_k$;
  Order;
  Reflect;
  Expand;
  Contract;
  Shrink;
end

Figure 3. Nelder-Mead Algorithm

following Algorithm:Nelder-Mead method (Fig. 3) is applied for the hyperparameter tuning problem in support vector machine modelling. This method also finds better hyperparameter settings reliably for support vector machines.

C. Configuration of SDN Controller

This paper used two IP based cameras connected Raspberry Pi 3b+ as SDN based camera and input to SDN controller based on Raspberry Pi 3b. And then the output will lead to application layer (in this case- OpenCV). All cameras are set up on each entrance to catch up the people in events.

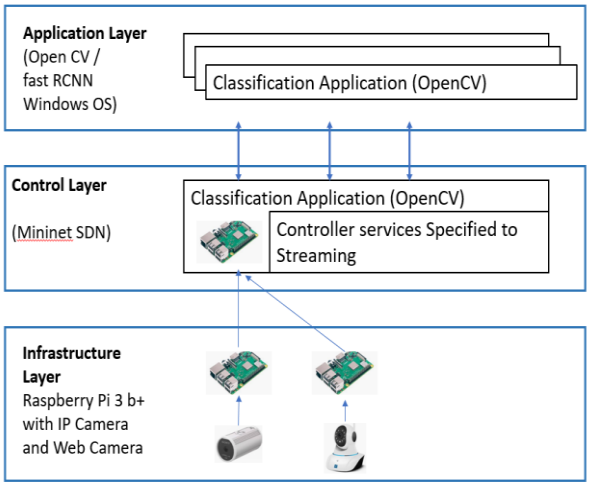

Figure 4. SDN basic configuration System

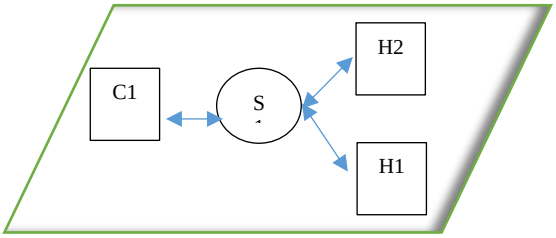

Figure 5. SDN basic configuration System

In a software-defined network, there are two types of architectures, physically distributed multi-controller architecture and physically centralized, just in the case of the physically distributed network. The system used a physically centralized controller at each pair of critical point for the underlying layer, there is just one controller that manage the specified video port over the whole network. The paper set SDN network with 2 hosts (h1, h2), 1 SDN switch (s1) and Controller (c1). POX SDN based controller is set up as centralized controller and Mininet is used as network emulator over Raspibian OS and emulate overall OpenFlow network on the overall system [10]. IP configuration for H1 and H2 is 10.0.0.2 and 10.0.0.3 respectively and this work set openflow default port for controller as 6633. H1 and H2 are two Raspberry Pi 3B+ with bypass port forwarding to entire system application notebook via controller.

TABLE I. IP ASSIGN AND PORT FORWARDING IN SDN CONTROLLER

Devices / Description	**IP or Port**
Nodes	c0 h1 h2 s1
Raspberry Pi with camera (h1)	10.0.0.2 (eth0)
Raspberry Pi with camera (h2)	10.0.0.3 (eth1)
SD Switch	s1-eth1:h1-eth0 s1-eth2:h2-eth0
Port forwarding (video)	8081

The above Table 1 shows the configuration set up for each node in the system OpenFlow.

IV. DATASETS AND TRAINING

This paper defined Gender in two classes – Male (0), Female (1) and trained in OpenCV to get configuration with two different Datasets, IMDB and Asia Image Dataset. Moreover, head position and facial expression had not considered in this system. Age class division did not follow equal interval, and this system defined as divided ten classes for Age- (0-2), (3-6), (7-13), (14-20), (21-24), (25-32), (33-37),(38-43), (48-53) and (54-70). In each dataset, the image file with metadata file included as following format - DOB, phototaken, full_filepath, gender and age group.

All fast R-CNNs are trained with openCV2 and additional hyperparameters are minimize the objective function by repeating its evaluation vertex. To set and generate initial parameters randomly, then perform optimization for up to 600 evaluations (included initialization). The system trained all two Datasets IMDB and Asia Image Datasets. Since IMDB original datasets has 460723, this system used IMDB wiki images as 3209 facial images while the second, Asia Image Dataset, is from Tsinghua Science and Technology 2019 with 13322 images [7] as Fig. 6.

In training process, the system divided the dataset into five sets, train the network with four sets, and test it with one set. Note that these processes require significant calculation time; thus, in the optimization process, cross validations are not performed. The system performs cross validation for only the optimal solution among the optimal solutions of all methods.

As per training time is too high, the system used to run over Open CV2 with GPU 640 Cores (GeForce GTX 1050 (2GB)) on desktop with 2 days.

V. EXPERIMENTAL RESULT AND FINDINGS

This system found that the classification on and accuracy of detection for Myanmar video streaming, used current Myanmar Idol season 4, as benchmark and accuracy rate is higher with hybrid hyperparameters trained system as Fig. 7. and Fig. 8.

VI. CONCLUSIONS AND FUTURE WORKS

In this system presented gender classification method based on fast R-CNN with hyperparameter optimizations and experimental results provided. The major contributions are twofold: (1) organizing Myanmar Image datasets for gender classification; (2) proposing new hyperparameter optimization configuring model for Myanmar people gender classification with competitive performance on the Myanmar Image Dataset and Asia Image dataset. To get better results, Myanmar Image Dataset with more images. Many tests and evaluation along with the reliability of Software-defined network based configuration and applications have many challenges. These challenges include scaling to ability to adjust easily between prototype and test environments.

Figure 6. IMDB Datasets and Asia Image Datasets

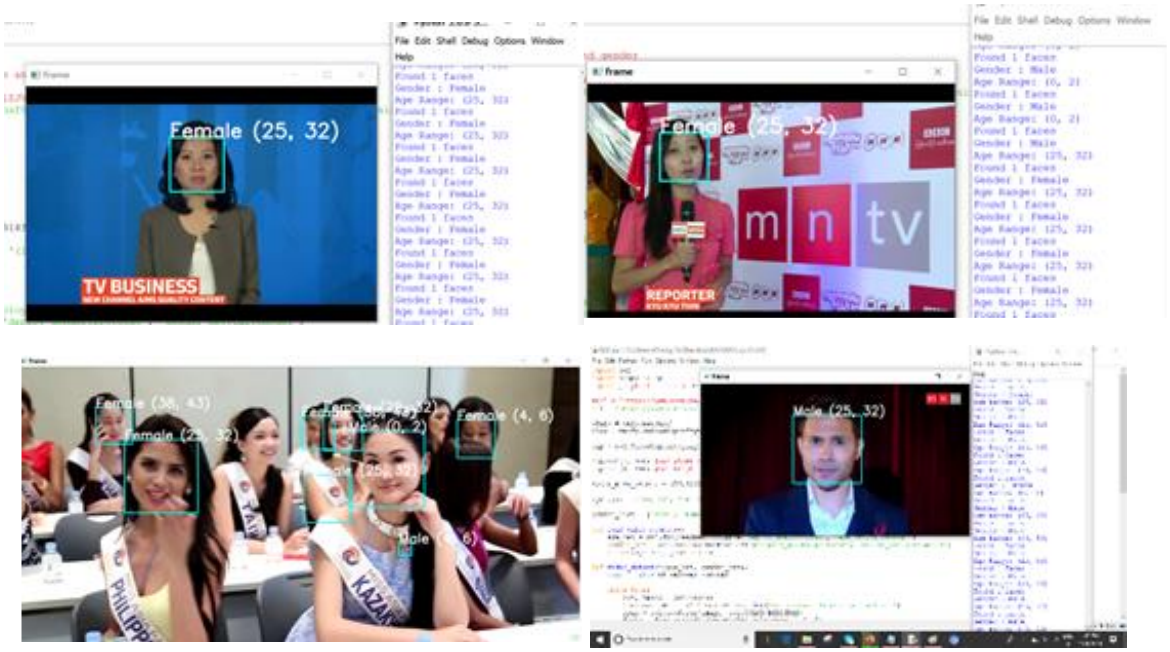

Figure 7. Experiment results

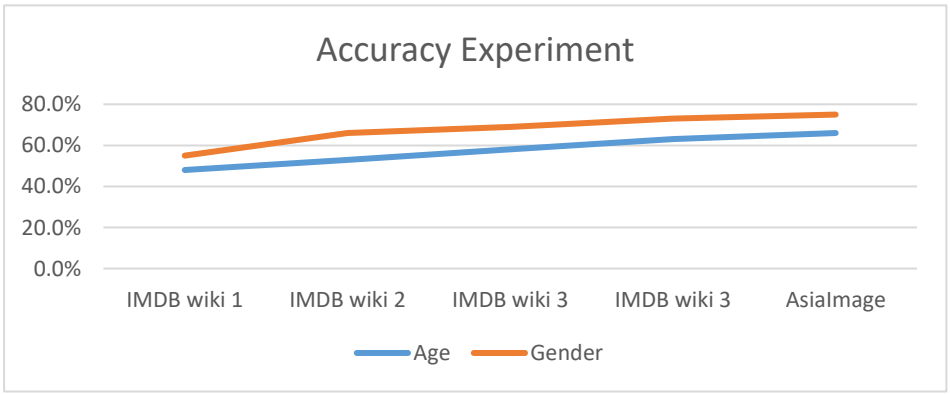

Figure 8. Accuracy Experiment

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Software Engineering and Modeling

Consequences of Dependent and Independent Variables based on Acceptance Test Suite Metric Using Test Driven Development Approach

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Abstract

The fundamental of software development was Test-Driven Development but the individual tests must be carried out previously the production code. To research, the consequence of test-driven development on product code quality and developer productivity was the destination of this paper. This system builds the acceptance test suite metric of regression analysis to assess the impact of the process on dependent variables and independent variables. This paper's results observed the positive effect of external quality over function of the number tests, and slightly decrease the effect of developer productivity over function of the number of tests. TDD can affect advance software products' quality, also mend programmers' productivity. TDD undertook to help the delivery of high-quality products, both operational (fewer bugs) and technical perspective (cleaner code) while improving developers' productivity. TDD affects to less defects and fewer debugging period which correct code can be certified by writing tests first and thus serving the developer get a finer understanding of the software requirements.

Keywords-- *Test-Driven development, Unit test, no: of tests, External Quality, Developer Productivity*

I. INTRODUCTION

By driving from Extreme Programming (XP) and the primary of the Agile Platform, the foundation fragment of the agile code development approach was Test-driven development (TDD). The possibility of TDD describes various positive effects. TDD isn't a testing approach, yet rather a development and design method in which the tests are composed before the production code. During the implementation, the tests are added step by step and when the test is passed, the code is refactored to improve the inside structure of the code, without changing its outside behavior. TDD cycle is iterated until the whole functionality is performed. An automated segment of code was a unit test that applied a part of work in the system and a unique idea about the execution of that part of work. For each little function of an application, TDD begins with designing and developing tests. First, the test is created that distinguishes and approves what the code will do in the TDD approach. Make the code and after that test in the typical testing process. The developer can be self- assurance that code refactoring is not destroyed any existing functionality for re-executing the test cases. Before the actual development of the application, TDD is a process of evolving and running automated tests. To create higher code quality, developers can motivate by coding standards, analyzing code automatically, doing code reviews and refactoring legacy code. For bugs and defects count, the system works by testing and debugging.

This paper is structured as follows. The issue of Test-Driven Development initiated in Section (1). The obviousness of the test numbers, quality of external code and, developer product on test-driven development (TDD) expressed in Section (2). Section (3) describes related work. Section (4) presents a framework of test-driven development. The contribution of the relation of the test numbers, quality of external code, and developer product is described in Section (5). Next, observational analysis of the proposed system is discussed in Section (6). Section (7) expresses compatibility of results. Finally, Section (8) concludes this paper.

II. OBJECTIVES

One of the approaches of software progression was test-driven development. In recent years, this approach has become familiar in the industry as a requirements specification method. Before code development, developers encourage to compose tests. TDD is provided to carry the code clearer, simple and

bug-free. The goal of the proposed system analyzes the consequence of dependent variables and independent variables on TDD**.** It observes the nature of the interaction between test numbers (TEST), quality of external code (QLTY), and the relation between the test numbers (TEST) and developers' product (PROD). This decreases the fault of enhanced software either instantly or in the long run. The benefits of TDD enhanced software quality and speed up the testing process. This approach aims more productive and make fewer efforts per line of code. By decreasing code complexity supporting, the proposed system validates the exactness of all codes and allows developers assurance. It is used persistently over time and motivates developers to create higher code quality.

III. RELATED WORK

In [5], Authors (Y. Rafique and V. B Misic) described on "The effects of test-driven development on external quality and productivity: A meta-analysis". Authors reported that TDD improves external quality when compared to a waterfall approach. However, this improvement is not strong. Further, TDD becomes disadvantageous for the subset containing only academic studies in which it is compared to an iterative, test-last (ITL) process instead of a waterfall approach. This result suggests that sequencing might have a negative effect on external quality, which we haven't observed. Productivity results are more inconclusive in that the authors report a small productivity hit for TDD when comparing TDD with waterfall but the effect, even though still small, is reserved when ITL is compared with TDD.

In [14], Authors (H. Munir, K. Wnuk, K. Petersen, and M. Moayyed) proposed on "An experimental evaluation of test-driven development vs. test-last development with industry professionals". The authors were developed that it intended to compare the effect performed by TDD and TLD (Test-Last Development) on the quality of internal and external code, and developer's product. For this aim, 7 user stories' a programming exercise was carried out. The results of the analysis by the approved test cases' number: McCabe's Cyclomatic complexity, branch coverage, the no: of code lines person/hour, and user stories' number described person/ hour. The tests expressed fewer significant enhancements in accept of TDD, by reducing the defects. In terms of productivity, the tests indicate that TDD than TLD slightly decrease average productivity.

In [15], Authors (M. Moayyed, H. Munir, and K. Petersen) described on "Considering rigor and relevance when evaluating test-driven development: A systematic review". Authors were developed that the primary studies are considered together; however, the nine better-rigors, better-relevance studies describe that TDD enhances quality of external code, while developer' product is not effected. The 21 studies in the alike classification of the basis analyze and this replication are ambiguous both results.

IV. BACKGROUND THEORY

Test-Driven Development is a coding technique. TDD accelerates the before time development of tests, at the time alternates are accepted and improved with functional components. Kent Beck invents Test-Driven Development applies to a form of programming although three actions are exactly interlinked. Three activities are Coding, Testing, and Design. At first, its key idea is to execute early initial tests for the code, must be actualized but the accurate feature of it used. One of the features of software system requirement is tackled subtask or user stories, which are designed to easily express and understand. These can be easy to change by the end-user as they like during the project's handle time.

A. Test-Driven Development

The TDD process is expressed in Figure 1, and includes the consecutively steps:

1. Write only one test-case
2. Run or perform this test-case. If this test-case fails, go to step 3. If the test-case succeeds, go to step 1.
3. Refactor the performance to get the elementary design possible.
4. Enable the minimal code to do the test-case run.
5. Run the test-case again. If it fails again, go to step 3. If the test-case succeeds, go to step 5.
6. Again, run the test-case, to certify that the refactored application until passes the test-case. If the test-case fails, go to step 3. If the test-case passes, go to step 1, if there are still requirements, left in the specification.

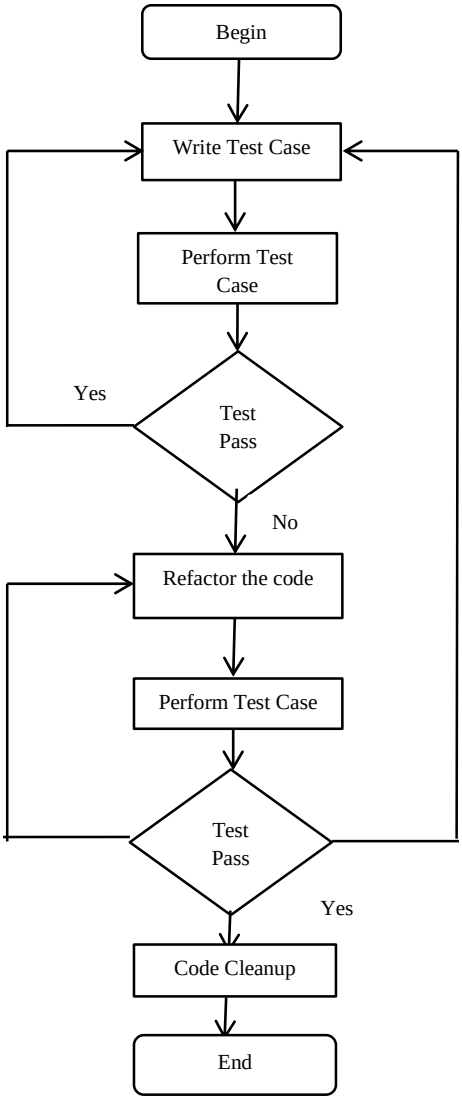

Figure 1: Test-Driven Development flow

V. CONTRIBUTION

First stage, the original study of Test-Driven Development has beneficial effects on the number of unit-test written by the developers, the external code quality and the developers' productivity. In the second stage, the authors studied the correlation between the number of tests, the external code quality, and productivity. TDD approach encourages developers to write more tests and is a positive correlation between the number of tests, quality, and productivity, and then TDD would improve the overall quality and productivity. The related work observed, if code quality has a positive effect, productivity has a negative effect, if productivity has a positive effect, code quality has a negative effect. The proposed work discovered, if code quality has a positive effect, productivity has a slightly decrease effect, if productivity has a positive effect, code quality has fewer reduced effect.

VI. PROPOSED SYSTEM

In this proposed method the acceptance test suite metric of regression analysis uses to measure the no: of test numbers, quality of external code, and developer product.

A. Research Questions

This system concentrates to evaluate two outcomes on the following system: external code quality and developer productivity.

RQ1 (RQ-QLTY): Does a higher number of tests indicate higher quality?

RQ2 (RQ-PROD): Does a higher number of tests indicate higher developer productivity?

The notion of external code quality in RQ-QLTY and productivity in RQ-PROD are based on the acceptance test suite metric of regression analysis.

B. Method

In the proposed system, the acceptance test suite metric is used by analyzing to explore possible interactions such as number of tests, external code quality, and developer productivity. The acceptance test suite metric is a form of mathematical regression analysis. Regression analysis is used to investigate the relationship between two or more variables and estimate one variable based on the others. Regression analysis is a powerful statistical method that allows for analyzing the relationship between two or more outcome variables of interest. QLTY and PROD are the dependent variables. TEST is the independent variable. QLTY defined as the percentage of acceptance tests passed for the implemented tackled tasks. PROD measured as the percentage of implemented tackled tasks. Table 1 provides the raw data used in the assessment. To compute this low-level measure, an automated tool used by this system. The limited-time necessary to complete the task had an impact on the metric. In regression analysis, dependent variables are established on the vertical y-axis, while independent variables are established on the horizontal x-axis.

C. Test Numbers (TEST)

Test numbers (TEST) is identified as JUnit assert statement numbers inside the unit test suite written by the participants while tackling the task. The numbers of test development as a single JUnit assert statements. TEST assessed by the count of the JUnit test cases. TEST is a ratio measure in the range [0, ∞]. The formula for calculating TEST is defined as [10]:

TEST = no: of subtasks out of result the no: of input subtasks (1)

TEST = JUnit assert statement numbers inside the unit test suite

Table 1: Summary of acceptance tests used to calculate the metrics of Bowling Scorekeeper data-sets [7].

Task	Test	Assert
T1	3	3
T2	3	3
T3	2	2
T4	3	10
T5	5	5
T6	6	6
T7	8	8
T8	5	5
T9	5	5
T10	4	4
T11	2	2
T12	3	3
T13	2	2

D. External code quality

The metric for external quality QLTY based on the number of tackled subtasks *(#TST)* for a given task. A subtask as tackled assesses if at least one assert statement in the acceptance test suite associated with that subtask passes. QLTY is a proportion measure in the range 0 to 100.

The number of tackled subtasks (*#TST*) is defined as:

$$\#TST = \sum_{i=0}^{n} \begin{cases} 1 & Assert_i(Pass) > 0 \\ 0 & otherwise \end{cases} \quad (2)$$

#TST = the number of tackled subtasks

n = the total number of subtasks

The formula for measuring QLTY is defined as [8]:

$$QLTY = \frac{\sum_{i=0}^{\#TST} QLTY_i}{\#TST} \times 100 \quad (3)$$

$QLTY_i$ = the i^{th} tackled subtask's quality

Where $QLTY_i$ is the quality of the i^{th} tackled subtask and $QLTY_i$ is defined as:

$$QLTY_i = \frac{\#Assert_i(Pass)}{\#Assert_i(All)} \quad (4)$$

#Assert$_i$ (Pass) = the number of JUnit assertions passing in the acceptance test suite associated with the i^{th} subtask

#Assert$_i$ (All) = the total number of JUnit assertions in the acceptance test suite associated with the i^{th} subtask

For example, supposing that the thirteen tackled subtasks (*#TST* = 13) assessed by a person, this denotes that the thirteen tackled subtasks pass more than one assert statement in the test suite. Assume us that the acceptance test of the first analyzed tackled task contains 3 assertions, out of results of three are passing. The acceptance tests of the fourth tackled task contain 10 assertions, out of results of three are passing and so on.

Table 2: Solution of QLTY

Task	Test	Assert	QLTY
T1	3	3	1
T2	3	3	1
T3	2	2	1
T4	3	10	0.3
T5	5	5	1
T6	6	6	1
T7	8	8	1
T8	5	5	1
T9	5	5	1
T10	4	4	1
T11	2	2	1
T12	3	3	1
T13	2	2	1
	51	**58**	**95**

i.e. ($QLTY_4 = \frac{\#Assert_4(Pass)}{\#ASSERT_4(All)} = \frac{3}{10} = 0.3$)

($QLTY = \frac{\sum_{i=1}^{\#TST} QLTYi}{\#TST} \times 100$

$= \frac{1+1+1+0.3+1+1+1+1+1+1+1+1+1}{13} \times 100 = 95$)

E. Productivity

The productivity metric (PROD) expresses the amount of work effectively carried out by the subjects. PROD is a proportion measure in the range 0 to100. The metric of PROD is computed as follows [8]:

$$PROD = \frac{\#Assert(Pass)}{\#Assert(All)} \times 100 \quad (5)$$

For sample, assume a tacked task with all of 58 assert statements enabled by a person in a test suite. After compiling, the person's outcome 51 asserts statements are passing.

i.e. ($PROD = \frac{\#Assert(Pass)}{\#Assert(All)} \times 100 = \frac{51}{58} \times 100 = 88$)

F. Assessment

The image below is a scatter plot**.** Scatter plots are used when this paper want to show the relationship between two variables. Scatter plots are known as relationship plots because they show how two variables are interrelated. This analytical tool is most often applied to show data correlation between two variables. This system expects that the regression assessment of the tackled task compiled from the quality of external code on test numbers by TDD responds positively to questions RQ1. In the same way, this system expects that the regression analysis of the tackled task compiled from the developer product on test numbers by TDD responds slightly decrease to questions RQ2.

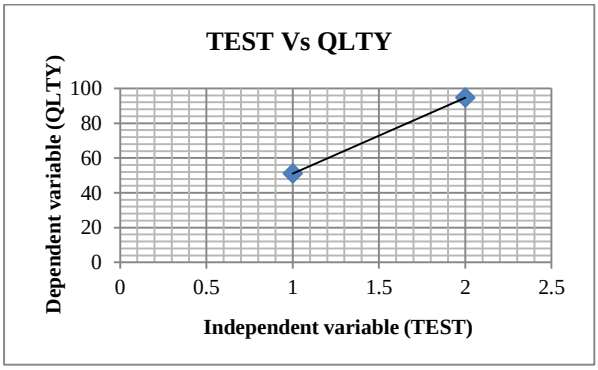

Figure 2: QLTY is on the function of TEST

In figure 2, the external code quality on the test numbers is improved by measuring the acceptance test suite metric of quality (QLTY).

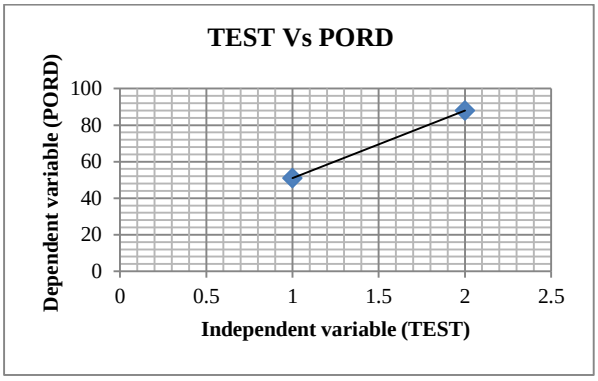

Figure 3: PROD is on the function of TEST

In figure 3, the developer's productivity over the test numbers is slightly decreased by measuring the acceptance test suite metric of productivity (PROD).

VII. COMPATIBILITY OF RESULT

In this portion, this paper presents the outcomes acceptance test suite metric of regression analysis. Further, a significant relation between TEST and QLTY, as expressed in *RQ1*, with a positive was found. Hence, scatter plot figure 2 is an arithmetically expressive relationship between the number of tests and external code quality. Additionally, a significant relation between *TEST* and PORD, as expressed in *RQ2*, with a somewhat down was found. So, scatterplot figure 3 is an arithmetically expressive correlation between the number of tests and programmer productivity. In this study, the number of tests is a good predictor for TDD programmer productivity. Consequently, developer product on the test numbers becomes lightly diminishment and quality of external code on the test numbers becomes improvement. Development time is relatively high in the proposed work. It nearly takes as much as 16% more time than that of related work. Proposed work decreases the maintenance cost and overall increases the productivity. There are as many as 52% more test cases as of the related work. The related work codes have a relatively small size. The inclusion of many more test cases in the proposed work increases the size of the code. The related work codes are simpler. The cyclomatic complexity of the related work is relatively smaller. The proposed work is relatively complex.

VIII. CONCLUSION

Development time is relatively high in the proposed work. It nearly takes as much as 16% more time than that of related work. Proposed work decreases the maintenance cost and overall increases the productivity. There are as many as 52% more test cases as of the related work. The related work codes have a relatively small size. The inclusion of many more test cases in the proposed work increases the size of the code. The related work codes are simpler. The cyclomatic complexity of the related work is relatively smaller. The proposed work is relatively complex.

This approach allows thorough unit testing which enhances the quality of the software and advances customer satisfaction. They help with retaining and varying the code. Moreover, the number of acceptance test cases passed and number defects found through static code analysis are used to measure the external code quality. All these measures are consistent with the studies and will be considered as standard measures. When this proposed system assesses the acceptance test suite metric of regression analysis, the result of developer productivity over the number of tests is fewer decreased and the result of external code quality over the number of tests is increased in giving a fixed time-frame.

ACKNOWLEDGMENTS

This research paper is partially supported by academic studies. Professionals were fit to implement more effective with test-driven development. Furthermore, this proposed system observes that the measurement reveal different aspects of a development approach in academic studies.

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Determining Spatial and Temporal Changes of Water Quality in Hlaing River using Principal Component Analysis

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Abstract

Water quality of a River is important as it is used for drinking, domestic purpose and agriculture. Hlaing River is one of the most important rivers in Yangon Region. The River plays an important role to supply water transportation for Hlaingtharyar, Insein and Htantapin Townships. The objective of the paper is to assess the change of water quality in Hlaing River in the wet and dry seasons. The secondary data used in the study are monthly mean temperature in Yangon (Average) between 2007 and 2016 and monthly rainfalls in Yangon (Average) between 2007 and 2016. Primary data are collected for sampling points. 8 numbers of samples were collected at different selective sampling points. A number of physiochemical water quality parameters including Biochemical Oxygen Demand (BOD), Chemical Oxygen Demand (COD) and Total Suspended Solid (TSS) are tested. Principal component analysis in SPSS is used to know the correlations of the variables and to determine how many important components are present in the data. Sample points located at the same river and nearer places are in same component.

According to the water quality surveys, the water quality of Hlaing River met the level of Class (III) of the Interim National Water Quality Standards of Malaysia (for Water supply) in wet season because of dilution process at that time and in dry season some part of Hlaing River met the level of Class (IV) of the Interim National Water Quality Standard of Malaysia (for irrigation purpose).

Keywords: *Hlaing River, spatial and temporal changes, industrial zones, dilution, irrigation*

I. INTRODUCTION

This paper is assessing water quality of Hlaing river which is mainly use of transportation purposes in Yangon Region. There are a lot of national and international standards of water quality assessment. In this paper the researchers classify and assess the result of water quality by using the Interim National Water Quality Standards of Malaysia (INWQS), a set of standards derived based on beneficial uses of water. As Myanmar and Malaysia are consisting of South East Asia countries, they have similar topography and environmental conditions.

Currently, Myanmar has already set up emission standards. However, it is still need to lay down the standards for ambient air quality and drinking water quality standards. In the absence of those national standards, the Ministry of Natural Resources and Environmental Conservation (MONREC) stated that internationally accepted environmental standards of World Health Organization (WHO) guidelines, to be adopted for any environmental assessment.

The water quality survey from "Project on capacity development in basic water environment management and Environmental Impact Assessment (EIA) system in Myanmar" is originally composed of total five-time surveys: in February 2016, June-July 2016, January 2017, June-July 2017 and February 2018 in order to cover the dry season and rainy season in two years, 2016 to 2017. In the water quality surveys, 21 parameters are collected and measured for water quality analysis. The assessment of project paper is to analyze the resulted data from water quality surveys which was implemented during 5 periods in 3 years.

This paper aims to analyze the current status of the Hlaing River through the on-site measurement and laboratory analysis.

II. DATA AND METHODOLOGY

A. Guideline levels and Classification

Ministry of Health and Sports (MOHS) and MONREC enacted Drinking water quality standard and National Environmental Emission Guideline for water. In Myanmar, there is no surface water quality standard. So, the researcher uses INWQS for surface water in order to compare with the results.

B. Sampling Parameters

For the water quality surveys the parameters are BOD, COD and TSS. In this project paper, these will be assessed and mentioned by using ArcGIS.

Table 1 shows the Interim National Water Quality Standards of Malaysia for surface water and the class mentioned in table are definitions of specific use of water quality with limit values.

Table I: Limit value and class of surface water quality parameters according to the Interim National Water Quality Standards of Malaysia (INWQS)

PARAM-ETER	UNIT	LIMIT VALUE and CLASS			
		IIB	III	IV	V
Biochemical Oxygen Demand (BOD)	mg/l	3	6	12	> 12
Chemical Oxygen Demand (COD)	mg/l	25	50	100	> 100
Total Suspended Solid (TSS)	mg/l	50	150	300	> 300

Source: INWQS

Note: Class IIB is recreational use with body contact,
Class III is Water supply – Extensive treatment required
and Common of economic value, and tolerant species; livestock drinking,
Class IV is irrigation purpose, and
Class V is none of the above.

C. Sampling method

Water samples were collected three times and then mixed in bucket. Then, take off 100 ml with sample bottle and send to the laboratory. Surface water was taken directly by a sampling bottle or using a plastic sampling bucket. In addition, Van Dorn Water Sampler was used to collect a water sample from a specific depth as needed.

D. Laboratory analysis method

BOD is analyzed by using Respirometric method (HACH Method 10099) in DOWA Eco-system-Myanmar Co., Ltd.

The researchers analyzed COD Cr with the Japanese Standard JIS K0102 (2016) 20.1 which is developed by OSUMI CO., Ltd.

TSS is analyzed by OSUMI CO., Ltd. in order to the method of Environment Agency Notification No. 59, 1971.

III. RESULT AND DISCUSSION

A. Location of the Study Area

Hlaing River is the middle section of a tributary of Ayeyarwaddy River, and its name evolves as Myit Ma Kha River, Hlaing River, and Yangon River as it flows down toward the sea. The length of Hlaing River is around 110 km-long and the starting point is the boundary between Yangon Region and Bago Region, and the ending point is before confluence of Pan Hlaing River between Hlaing Township and Hlaing Thar Yar Township in Yangon City (Figure 1).

The project mainly focuses on the downstream area of Hlaing River located in Yangon City because there are many industrial zones and the water quality may be deteriorated by the impact of wastewater from industrial zones.

This paper were be evaluated the water quality status of Hliang River which is based on the collected information and the data of **“**Project on capacity development in basic water environment management and EIA system in Myanmar” which is bilateral technical cooperation project between Myanmar and Japan during the project implement from June, 2015 to May 2018 in Yangon Region.

Figure 1. Location Map of the Study Area, Hlaing River

B. Spatial and Temporal Variations of Water Quality in Hlaing River

Basin map of Hlaing River with locations of industrial zones in the river basin is shown in figure 2. Hlaing River Basin is long in North-South direction, and the industrial zones are located in the most downstream area. Hlaing River connects with Yangon River and drains to the Andaman Sea.

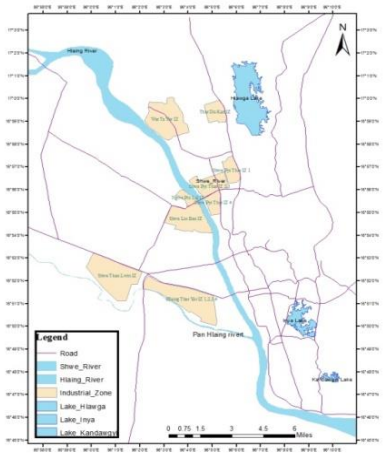


Figure 2. Hlaing River Basin and Locations of Industrial Zones

1): Comparison and Analysis of BOD in Dry and Wet Season

At the points of Hlaing - 1 and Pan Hlaing - 2, the value of BOD for all sampling points meets the level of Class V of the Interim National Water Quality Standard of Malaysia because the untreated waste water discharge along the Hlaing River was high at dry season. In the case of wet season, the values of BOD along the Hlaing River and Pan Hlaing River meet the level of Class IIB of INWQS for recreational use with body contact. It can be assumed that the dilution of heavy rain may have an effect on the value of BOD in river water quality and the suspension of distillery factories were happened at that time (Figure 3).

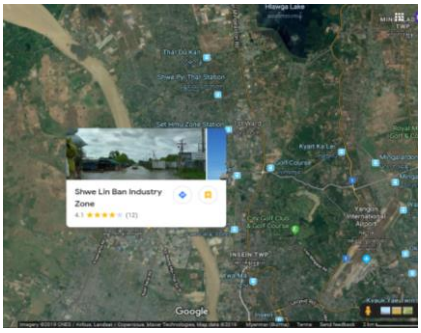
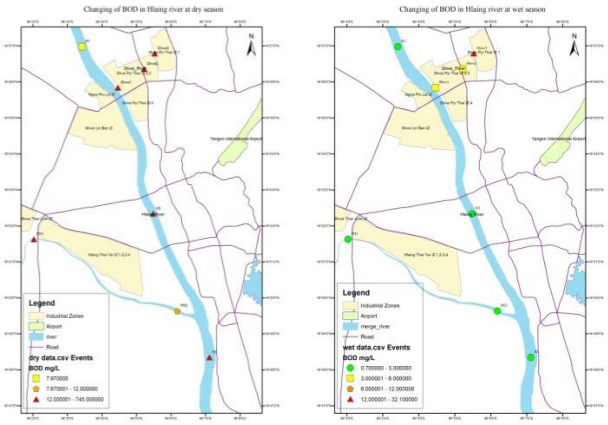


Figure 3. Location of Industrial Zones in the Study Area

According to the results of wet season, in Shwe creek the value of BOD at the points of Shwe - 1 and Shwe - 2 meet the level of Class III of INWQS with the purpose of water supply and fishery in the requirement of extensive treatment while the point of Shwe - 3 meets the level of Class V of INWQS which is located at Shwe Pyi Thar IZ area, may be polluted by the industrial waste water.

The following figure (4) shows comparison of changing BOD along Hlaing River Basin in dry and wet season.



(a) (b)

Figure 4. Comparison of Changing BOD along Hlaing River Basin in Dry (a) and Wet Seasons (b)

2): Comparison and Analysis of COD in Dry and Wet Season

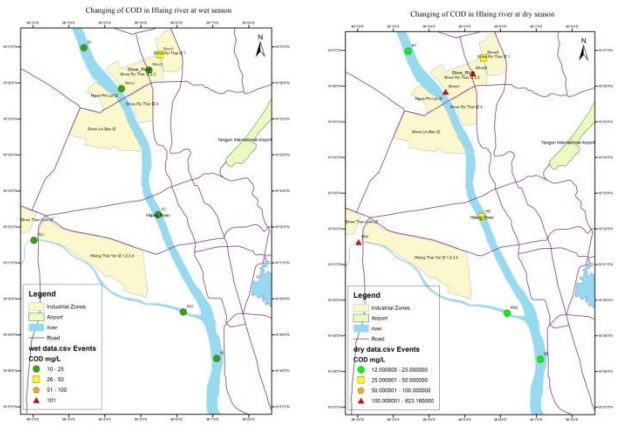
In the case of wet season, the value of COD along the Hlaing River, Pan Hlaing River and some points at the sub stream of Hlaing River, especially Shwe - 1 and Shwe – 2, meet the level of Class IIB because the dilution of heavy rain and the suspension of distillery factories may effect on the value of COD in river water quality. The upper stream of creek path (Shwe-3) meets the level of Class III of INWQS with the purpose of water supply and fishery in the

requirement of extensive treatment. But the water quality was relatively different from all sampling points which can be caused by some other pollution activities of Shwe Pyi Thar IZ so that further investigation is needed near it.

In the dry season, the water quality at the middle point of Hlaing River (Hlaing - 2), was better than that of upstream and downstream points, which was indicated by the impact of Shwe creek in which the value of COD at the points of Shwe - 1 and Shwe - 2 meet the level of Class III and Shwe - 3 meets the level of Class V of INWQS.

In Pan Hlaing River that is flowing through the Hlaing Thar Yar IZ into Hlaing River, the water quality of Pan Hlaing - 1 meets the level of Class V of the INWQS at the dry season. The downstream of Pan Hlaing River (Pan Hlaing - 2) was within the level of Class IIB.

Figure 5 shows comparison of changing COD along Hlaing River Basin in dry and wet season.



(a) (b)

Figure 5. Comparison of Changing COD along Hlaing River Basin in Dry (a) and Wet Seasons (b)

3): Comparison and Analysis of TSS in Dry and Wet Season

In the dry season, the value of TSS at the points of Hlaing-1 meets the level of Class IV of the Interim National Water Quality Standard of Malaysia with the purpose for irrigation while the other two points of Hlaing-2 and Hlaing-3 meet the level of Class V of INWQS. The former can be assumed that there may have some other activities to happen erosion at the upstream of Hlaing River and the latter can be caused by increasing rate of erosion which is higher than that of the upstream of Hlaing River. It can be assumed that these two points is located at the downstream and near of industrial zones.

In the wet season, the value of TSS along the Hlaing River meets the level of Class V of INWQS because the rate of erosion and tidal fluctuation were relatively higher than the dry season. That is why the water color of Hlaing River was turned to brownish with the less transparency in color.

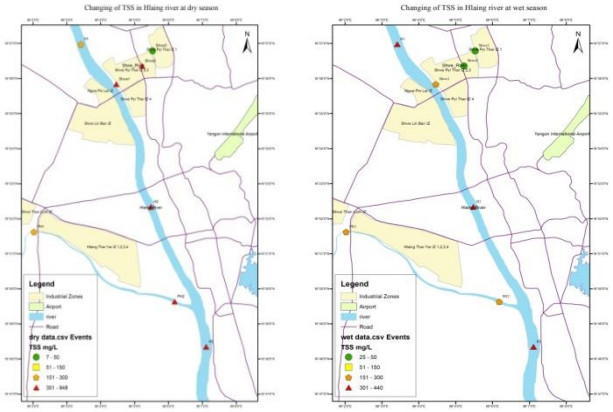
According to TSS results at the dry season, Pan Hlaing - 2 can be used for irrigation with the level of Class IV and Pan Hlaing -1 was within the level of Class V in INWQS. Pan Hlaing -1 and Pan Hlaing -2 meet the level of Class IV of INWQS at the wet season. It can be assumed that the downstream of Pan Hlaing is located at the downstream of industrial zones and their activities.

In the case of Shwe creek at the dry season, the value of TSS for Shwe - 1, meets the level of Class IIB while Shwe - 2 and Shwe - 3 meet the level of Class V according to INWQS. It can be assumed that there are no erosion activities near Shwe - 3 point. The water quality of the other points at Shwe Creek passed through Shwe Pyi Thar IZ area which may be caused by industrial activities.

In the wet season, TSS value at the Shwe - 2 and Shwe - 3 points meet the level of Class IIB of INWQS but the downstream point Shwe - 1 meet the level of Class IV of INWQS. It can be assumed that the downstream of Pan Hlaing is located at the downstream which is passed through the Shwe Pyi Thar industrial zone.

The differences between the dry and wet season are that the activities in Hliang River Basin and heavy rain can cause the erosion which is resulted in changing the value of TSS.

The following figure shows comparison of changing TSS along Hlaing River Basin in dry and wet season.



(a) (b)

Figure 6. Comparison of Changing TSS along Hlaing River Basin in Dry (a) and Wet Seasons (b)

C. Principal Component Analysis

To know 8 sample points in Hlaing River have similar patterns of responses i.e. do these points hang together to create construct? Principal component analysis (PCA) can explain the interrelationships among the variables of 8 sample points.

The main idea of PCA is to reduce the dimensionality of a data set consisting of many variables correlated with each other, either heavily or lightly, while retaining the variation present in the dataset, up to the maximum extent.

The 1st principal component retains maximum variation that was present in the original components. The principal components are the eigenvectors of a covariance matrix, and hence they are orthogonal.[5]

This paper studies water quality changes of 8 places. Each place is described by its attributes like BOD and COD contents according to the time etc. However, the complicated data will arise because many of them measure related places and variables. Therefore, PCA will do in this case is summarizing each station in the dataset with less characteristics.

The following table shows correlations in SPSS analyze by using bivariate. Form the table the most items have some correlation with each other ranging from r = -.146 for Shwe Creek 1 and Hlaing 2, “these are the nearest places” to r = 1 for the sample points in the same river. Due to relatively high correlations among items, this would be a good component. Fewer interrelationships variables can be broken up into multiple components such as sample points in Shwe Creek and Hlaing River.

Eigenvalue represents the total amount of variance. The total variance is made up to common variance and unique variance, and unique variance is composed of specific and error variance. Component 1 and 2 have the highest percentage of variance the former is 69.186% of variance in initial Eigenvalues and the latter is 30.669%and component 3 is 0.1% (Table 3).

To select the optimal number of components that are smaller than the total number of items, scree plot is performed which plots the eigenvalue by the component number. To choose components, one criterion is to have eigenvalues greater than 1. In Figure 7 the first two components have an eigenvalue greater than1. The first component has the highest total variance and the least components are at the last of the plot. Component 2 is making the joint position. The researcher selects two components on the basis of the scree plot.

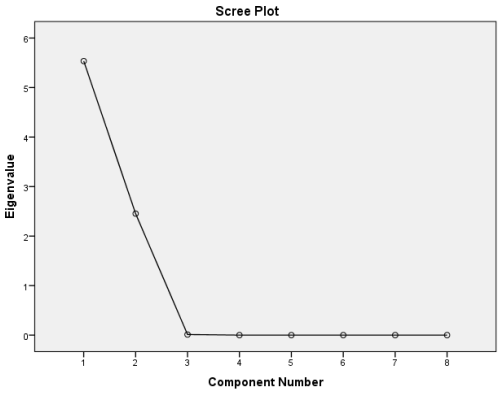


Figure 7. Scree Plot for Components

Component Matrix table contains component loadings, which are the correlations between the variable and the component. Hlaing 1, Hlaing 2, Hlaing 3, Pan Hlaing 1, Pan Hlaing 2 and Shwe 3 load heavily on the first component and the other two on the second component. This makes the output easier to read by removing the clutter of low correlations that are probably not meaningful anyway. Pan Hlaing 1 and Shwe 3 has fairly correlation with component 2. It can be clearly seen in figure 8.

Table IV. Component Matrix

Component Matrix[a]

	Component	
	1	2
Hlaing-1	.970	-.244
Hlaing-2	.963	-.269
Hlaing-3	.966	-.258
Pan Hlaing-1	.903	.426
Pan Hlaing-2	.962	-.271
Shwe-1	.128	.990
Shwe-2	.236	.971
Shwe-3	.959	.278

Extraction Method: Principal Component Analysis.

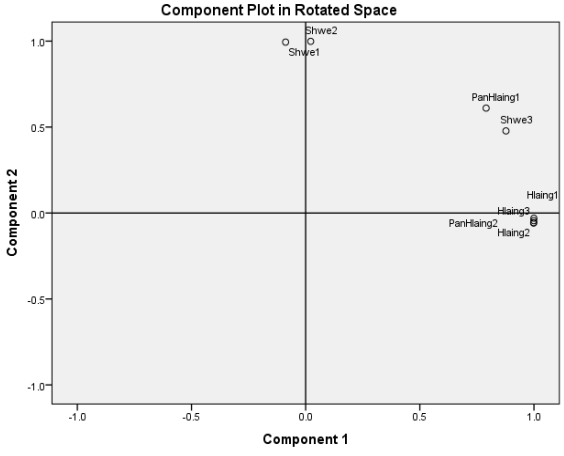


Figure 8. Component Plot in Rotated Space

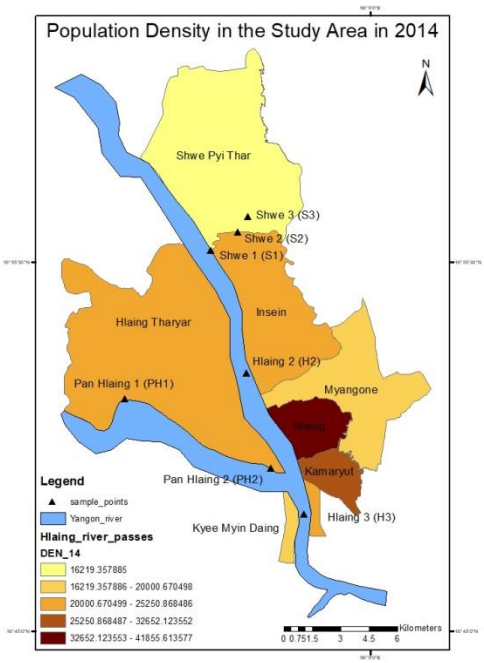


Figure 9. Population density and Sample Points

The study area is located in the western part of Yangon River. Rapid urbanization is pronounced in these large townships – Shwepyithar, Hlaingtharyar, Insein and Mayangon. Shwelinban industrial zone and Shwepyithar industrial zones are established in these townships. The consequences of human activities such as municipal wastewater, waste from the factories and urban drainage into the rivers effect on the water quality of the area. Chemical and biochemical contents in the water in dry season are more pronounced than wet season. Huge discharge of municipal wastewater into the river is one of the important factors.

IV. CONCLUSION

Hlaing River in Yangon Region has been selected as the study area because the impact of industrial zones is important for the quality of water. To examine the comparison of spatial and seasonal changes of surface water quality in Hlaing River physiochemical water quality parameters such as BOD, COD and TSS are tested.

The paper is applied on-site measurement and laboratory analysis result data of BOD, COD and TSS from 3rd and 4th water quality surveys which are mainly developed by the Project for Capacity Development in Basic Water Environment Management and EIA System.

The results of wet and dry season surveys were evaluated with respect to each sampling points of rivers and creek.

At the middle point of Shwe creek (Shwe-3), the water quality was significantly deteriorated, which was indicated by high level of COD and BOD. The result did not show marked deterioration of surface water quality in the flow direction of Hlaing River and Pan Hlaing River. The creek passing through the industrial zones and then flowing into Hlaing River, its water quality is significantly deteriorated by wastewater, which is indicated by low Dissolved Oxygen (DO) as well as high concentrations of COD and BOD.

In dry season, places located near the industrial zones have the high COD and the opposite true in wet season. Similarly, in dry season the content of BOD is high in nearly all places, but the value is less than 0.7 mg/l in wet season exception Shwe 3 point. In the study area, the content of BOD and COD in the water is high in dry season and low in wet.

Conduct a principal component analysis can determine how many important components are present in the data. It can highlight the correlation between the variables and components. Shwe 1 and Shwe 2 points (located near the Creek from Shwe Pyi Thar Industrial Zone to enter Hlaing River) have the same environmental background and Shwe 3 and Pan Hlaing 2 have the same situations such as located in the industrial zones. Hlaing 1, Hlaing 2, Hlaing 3 and Pan Hlaing 2 located along the rivers are one group. Therefore, PCA can point out the grouping of the variables on the plot.

In the Hlaing River of Yangon, the problem of various water discharged manner from industrial zones is one of the most important problems to protect water quality and its environment especially in Hlaingtharyar, Hlaing and Insein townships having the large number of population.

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Appendix

Table II. Correlation Matrix of Sampling Points in the Study Area

Correlation Matrix[a]

		Hlaing-1	Hlaing-2	Hlaing-3	Pan Hlaing-1	Pan Hlaing-2	Shwe-1	Shwe-2	Shwe-3
Correlation	Hlaing-1	1.000	1.000	1.000	.772	1.000	-.118	-.008	.862
	Hlaing-2	1.000	1.000	1.000	.755	1.000	-.144	-.034	.849
	Hlaing-3	1.000	1.000	1.000	.762	1.000	-.131	-.023	.856
	Pan Hlaing-1	.772	.755	.762	1.000	.753	.534	.629	.982
	Pan Hlaing-2	1.000	1.000	1.000	.753	1.000	-.146	-.037	.848
	Shwe-1	-.118	-.144	-.131	.534	-.146	1.000	.988	.401
	Shwe-2	-.008	-.034	-.023	.629	-.037	.988	1.000	.494
	Shwe-3	.862	.849	.856	.982	.848	.401	.494	1.000

a. This matrix is not positive definite.

Table III. Total Variance Explained

Total Variance Explained

Component	Initial Eigenvalues			Extraction Sums of Squared Loadings			Rotation Sums of Squared Loadings		
	Total	% of Variance	Cumulative %	Total	% of Variance	Cumulative %	Total	% of Variance	Cumulative %
1	5.535	69.186	69.186	5.535	69.186	69.186	5.393	67.408	67.408
2	2.453	30.669	99.854	2.453	30.669	99.854	2.596	32.446	99.854
3	.012	.146	100.000						
4	8.310E-16	1.039E-14	100.000						
5	1.940E-16	2.425E-15	100.000						
6	7.542E-17	9.428E-16	100.000						
7	-2.145E-16	-2.681E-15	100.000						
8	-6.207E-16	-7.759E-15	100.000						

Extraction Method: Principal Component Analysis.

Results of Laboratory Analysis

Table V: 3rd Water Quality Result

No	No	Unit	2	3	4	5	6	7	8	9
	Location name		H1	H2	H3	PH1	PH2	Shwe1	Shwe2	Shwe3
			Hlaing-1	Hlaing-2	Hlaing-3	Pan Hlaing-1	Pan Hlaing-2	Shwe-1	Shwe-2	Shwe-3
	Sampling Date		30/1/2017	30/1/2017	30/1/2017	30/1/2017	30/1/2017	30/1/2017	30/1/2017	30/1/2017
	Sampling Time		09:30	10:25	09:12	11:13	10:00	12:30	13:08	13:45
1.	BOD	mg/L	7.67	14.33	15.38	120.00	10.38	745.00	618.75	34.38
2.	Total Suspended Solids (TSS)	mg/L	192.00	948.00	434.00	241.00	554.00	396.00	426.00	75.00
3.	COD Cr	mg/L	12.50	33.34	18.76	159.77	16.67	823.18	781.50	37.51

Results of Laboratory Analysis

Table VI: 4th Water Quality Result

No	No	Unit	2	3	4	5	6	7	8	9
	Location name		H1	H2	H3	PH1	PH2	Shwe1	Shwe2	Shwe3
			Hlaing-1	Hlaing-2	Hlaing-3	Pan Hlaing-1	Pan Hlaing-2	Shwe-1	Shwe-2	Shwe-3
	Sampling Date		18/9/2017	18/9/2017	18/9/2017	18/9/2017	18/9/2017	19/9/2017	19/9/2017	19/9/2017
	Sampling Time		10:10	11:05	12:01	15:41	14:45	13:11	13:46	08:40
1.	BOD	mg/L	1.16	1.80	1.26	1.57	0.72	5.41	5.94	32.07
2.	Total Suspended Solids (TSS)	mg/L	390	440	330	230	290	180	27	25
3.	COD Cr	mg/L	14	14	10	14	12	19	22	44

Process Provenance-based Trust Management in Collaborative Fog Environment

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Abstract

With the increasing popularity and adoption of IoT technology, fog computing has been used as an advancement to cloud computing. Although trust management issues in cloud have been addressed, there are still very few studies in a fog area. Trust is needed for collaborating among fog nodes and trust can further improve the reliability by assisting in selecting the fog nodes to collaborate. To address this issue, we present a provenance based trust mechanism that traces the behavior of the process among fog nodes. Our approach adopts the completion rate and failure rate as the process provenance in trust scores of computing workload, especially obvious measures of trustworthiness. Simulation results demonstrate that the proposed system can effectively be used for collaboration in a fog environment.

Keywords: *process provenance, trust, collaborative, fog computing*

I. INTRODUCTION

Cloud computing is a computing paradigm that gathers and investigates real-time processing data above the internet [1]. However, as the substantial long-distance between cloud data centers and IoT edge devices, cloud computing suffers from traffic jams, significant end-to-end delay, communication charges and processing of massive amounts of data. Thus, to overwhelm these problems, fog computing has appeared as an improvement to the conventional cloud computing framework for providing geographical distributions, low latency and critical workload computation along with area awareness [2]. It affords workload computation, storage resources, and application services like a cloud. To protect interconnection among fog computing nodes, the following relations between these nodes are to be defended: (i) relations between fog computing nodes and IoT devices, and (ii) the relations among fog computing nodes. Collaborating among fog computing nodes is required to provide the subsequent amount of localized control, configuration and management, shared resources and computing power.

One of the ultimate goals of Fog computing is increasing the reliability of the network. Thus, the trust management mechanism is necessarily needed for efficient collaboration of fog computing nodes. Trust can also provide reliability by allowing assistance for selecting the fog nodes to cooperate. Employing trust management in fog computing environments will admit fog clients, resource-constrained devices, and fog nodes to forecast future behavior of each other. Based on the prediction of future behavior of fog devices, fog clients can select a fog node in the vicinity that will provide the best service. To predict the future event, fog nodes need to know how they work in the past to measure whether it can be trusted or not. Thus, provenance information is necessary for consideration of trust management.

Provenance is a procedure that traces the past interaction of a process or data item, beginning from its sources. Our approach is based on the concept of provenance, as provenance gives important evidence about the origin of the process or data item. Process provenance, gathered about the progress, traces the calling of services during a process's execution and enables recording of process flow and services in a collaborator environment. Consequently, process provenance is considered as one of the major requirements for trust establishment in the IoT devices and their attached fog nodes.

However, there were provenance-based trust models in wireless sensor networks [3] but a provenance-based trust model in fog environment has not been addressed properly. The studies in [6,7,8] show that feedback based trust models have been provided in fog nodes but malicious fog node can give a good recommendation to other malicious nodes.

To solve these issues, we propose the provenance-based trust mechanism in a fog environment. The main contribution of this work is to develop a direct trust approach based on the trust value of computing jobs among fog nodes, tracing the behavior of the process flow within fog nodes. We summarize the contributions of this paper as follows:

a) We develop a direct trust mechanism for fog collaboration which leads to efficient sharing of resources based on the process provenance.

b) We compute the trust value based on the computing history of jobs on each fog node.

c) We evaluate the trust among fog nodes according to the provenance based trust value of computing jobs.

The remainder of this paper is grouped as follows: Section 2 presents the related works. Section 3 presents how the direct trust approach with provenance in fog computing works. The method and design used for the provenance-based trust mechanism are described in Section 4. The simulation results are analyzed in Section 5 and finally, our conclusion and future work are drawn in Section 6.

II. RELATED WORD

Many of researcher groups are carrying out in the area of trust mechanism in fog and cloud computing environments.

In [3], the authors proposed a provenance-based trustworthiness assessment in sensor networks. They emphasized the use of unreliable data may lead to catastrophic failures as the main contribution of the provenance system. However, they intended only for the trust calculation framework of data elements and network devices based on the interdependency between data and network nodes, not for achieving trust based on process provenance that can accelerate the collaboration among fog nodes.

In [7], the authors proposed a reputation-based trust management framework that provided a group of process to provide Trust as a Service (TaaS), which include a new protocol to validate the integrity of feedbacks and protect the privacy of users. And it was a flexible and robust integrity model for estimating the integrity of trust feedbacks to cover cloud services access from the unauthorized user. Moreover, it implemented the decentralized trust management service. However, this system mainly depends on the user's feedback to get the reliability of cloud services, instead of calculating trust by themselves.

A Reliable and Lightweight Trust Computing Mechanism for IoT Edge Devices Based on Multi-Source Feedback Information Fusion was proposed in [8]. They used lightweight trust evaluating mechanisms for the collaboration of IoT edge devices. The multi-source feedback mechanisms are used for overall trust computing methods to get more trustworthy against bad-mouthing intrusions made by malicious recommenders. However, this study mostly focused on feedback mechanism that is based on third party recommender, so misleading feedbacks can lead the unauthorized user and waiting time to get the feedback will be an issue. Our proposed system, on the other hand, targets the direct trust framework among fog devices to avoid malicious recommenders.

In [9], the authors proposed ProvChain, a provenance framework to support the cloud environment. They adopted a blockchain mechanism to protect the privacy and unauthorized access of data provenance. The authors proposed a trust model for a cloud environment using secure data provenance in [10]. Their works had two parts to get assure provenance method: keeping the user privacy and historical data employing cryptographic mechanisms. These trust frameworks involved collecting and achieving historical data. However, these two studies mainly focused on a secure data provenance scheme in a cloud environment, not for process provenance in fog architecture. Our work varies from them by analyzing the trust among fog node to collaborate based on the process provenance.

A blockchain-based process provenance for cloud forensics was proposed in [15]. They proposed a process provenance, which affords validation of reality and privacy conservation for recording workflows and cryptography group signature. As a result, this study raises the trustworthiness of the chain of protection for cloud forensics. However, the privacy of their scheme is depending on the trustworthiness of the third-party organization. Our work differs from them by using the process provenance to build a trust model for fog environment. Our work leads to effective collaboration among fog nodes with the support of trusted fog devices.

III. BACKGROUND

A. Fog Computing Architecture

We employ the architecture of fog computing as shown in Fig. 1. It involved three layers: Cloud, Fog and Edge. The cloud layer exists at the top core layer and long distance from the IoT edge devices. The fog layer exists at the center and is closely connected to the IoT edge devices. The fog networks can be connected to the cloud server. Each IoT edge device is connected to a Fog node. Fog nodes are serving as the management layer between the cloud and edge devices. All communications such as Fog-to-Fog, Fog-to-Cloud, and Fog-to-Edge are back and forth communications.

1) *Cloud Layer*: Especially this layer serves as high-administration services for computing, storage, and processing. This layer also acts as the remote server and management control center that maintains a large amount of data and extremely complex processes. But they can store and process the non-critical computing jobs. The workloads from the IoT devices are delivered to the cloud through the fog network. Cloud servers grant inevitable and global coverage [12].

2) *Fog Layer*: This layer acts as a management layer between the cloud and the IoT edge devices. It involves communication among Fog devices. It affords geographical distributions, low latency and critical workload computation along with area awareness. Fog devices are also serving as a micro-data center for resilient storage. The services that can provide are distributed communications, workload computation, storage resources, and service. Comparison with the IoT edge devices, fog nodes have more repository to process the computing workload from the IoT edge devices. Otherwise, if the fog nodes require a much more complex and steady computation workload, this workload may be dispatched to the cloud servers. The fog nodes also act as bridges between the cloud and IoT edge devices. The fog devices can be relevant for fog cooperation. Collaborative and administrative policies are adopted on fog devices to provide management services. Fog collaboration can be accomplished by remote or local interconnection among them [12].

3) *Edge Layer*: It includes billions of heterogeneous IoT edge devices allowed with their unique identification, remote sensing, and communication capacities, such as mobile devices, smart cameras, sensors, and vehicles. IoT edge devices can connect many of the Fog nodes. These are very valuable and delayed to deliver all sensing data from end-device to cloud server. Edge devices cannot transfer important data to the cloud immediately. Thus, they are connecting to the fog device [12].

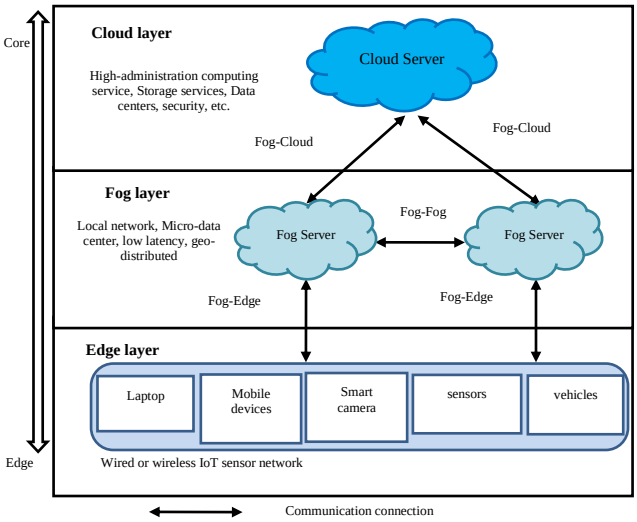

Figure 1. The architecture of Fog Computing

B. Trust in Fog Computing

Generally, trust shows a vital part in advance relations based on preceding interactions among fog computing nodes and IoT edge devices. The fog computing has a goal of increasing the reliability of the network. Thus, trust can provide reliability by allowing assistance for selecting the fog nodes to cooperate. Trust management mechanisms can be applied in fog computing architecture [5] to predict the future behavior of each other. With the prediction of future behavior of fog devices, client fog devices can choose a fog node in the proximity.

A fog computing node itself may be the malicious node to achieve the privacy and anonymity of end-users [14]. Also, these nodes must be an assurance for authorization, as the fog computing node carried out encrypted data and process. Thus, all fog nodes in the network have a certain trust level.

C. Process Provenance

Provenance indicates the derivation of origin data and processes that involve constructing an artifact. Provenance information is necessary to check the current information is trusted, how to organize different information origins and how to support authority to innovator when reusing information. It involved two parts: data provenance and process provenance. The data provenance [9] is

the origins of a sample of data that it takes place in a database. The process provenance is the recording of workflows that illustrates the service invoking within a process execution and permits the tracing of process and services in collaboratory environments.

Provenance in scientific workflow [13] is a record of the derivation history of scientific results. There are two types of provenance for scientific workflows: prospective and retrospective. Prospective provenance, takes an abstract workflow as a receipt for future data derivation. Retrospective provenance, takes past workflow execution and data derivation information, affords critical contextual information for the comprehensive analysis of scientific results. Our proposed system adopts the retrospective provenance to record the execution time of the computational tasks historically arrived at the application module because if we use prospective provenance, it is necessary to predict the future jobs' execution time.

IV. PROPOSED TRUST MECHANISM

A. Framework

The framework of the process provenance-based trust management approach consists of modules, namely, job history log, trust calculation module and collaboration module as shown in Fig. 2.

a) *Job History Log:* The start time and finish time of past computing jobs are recorded in the job history log module to calculate the execution time of jobs. When a user interacts with their application module to the fog service, fog nodes recorded the execution time of workload in each application module. As the fog node maintains the history interaction of computing jobs, the job history log for each fog node is saved and stored at the fog node.

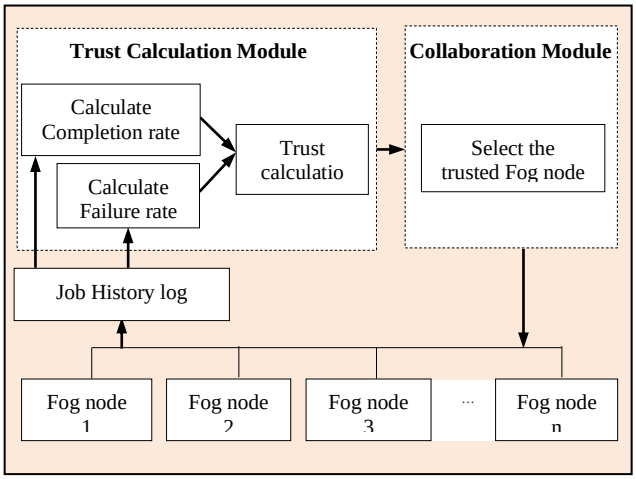

Figure 2. Framework

This job history log will be adopted by the trust calculation module.

b) Trust Calculation Module: This module calculates the trust value of each fog node. The trust value of each fog node i can be calculated as $T_i(\Delta t) = T_c(\Delta t) - T_f(\Delta t)$ where $T_c(\Delta t)$ is the trust value of job completion rate within deadline and $T_f(\Delta t)$ is the trust value of job failure rate and (Δt) is a time window of history interaction. To calculate the completion rate of jobs $T_c(\Delta t)$, compute the division of the number of completion time of jobs within deadline $J_{td}^{+}(\Delta t)$ and the total number of jobs. The job failure rate $T_f(\Delta t)$ can be computed from the division of the amount of job failure $J_{td}^{-}(\Delta t)$ as some serious abnormalities and the total number of jobs.

c) Collaboration Module: This module determines which fog nodes will be chosen to collaborate according to the trust value of each fog node. Therefore, it maintains the trust value series of each fog node, $T_S(\Delta t) = \{T_1, T_2, \ldots, T_n\}$. Then it decides to choose the fog node that has the highest trust value as trusted fog device for effective collaboration.

B. Process Provenance Trust Mechanism

The pseudo-code of our proposed trust mechanism is described in algorithm 1. Our proposed trust framework requires supporting long-term trust calculation for each fog device. Firstly, we take the list of fog computing nodes that are connected to the sensors and the job sets from sensors with a time slide (Δt) of past interactions, which is the massive number of past interaction records taken by our trust model. Then, we calculate the trust value of each fog node. According to the difference trust value of the job completion rate and job failure rate, we achieve the trust value of fog devices. The trust value calculation process is iterated for all fog devices in the system. According to the trust value series of fog devices, a fog device with maximum trust value will be chosen for collaboration.

Algorithm 1: Selection of Trusted Fog node for collaboration

1: **Input**: Fog nodes (F={f1, f2,…, fi,….., fn);
A job set (J = {j1, j2,…., jk,….., jm);
and time-series of history interaction
(Δt) for trust calculation;
2: **Output**: To calculate trusted fog, $T_F(\Delta t)$;
3: **Begin**
4: Calculate the trust value of each fog node
for (i = 1 to n) **do**

$T_i(\Delta t) = T_c(\Delta t) - T_f(\Delta t)$;

where $T_c(\Delta t) = \frac{J_{td}^{+}(\Delta t)}{\sum_{i=1}^{n} J_i(\Delta t)}$

//trust value of job completion rate

$T_f(\Delta t) = \frac{J_{td}^{-}(\Delta t)}{\sum_{i=1}^{n} J_i(\Delta t)}$

// trust value of job failure rate

end for

5: According to the trust value of each fog node, we can obtain the trust value series,

$T_S(\Delta t) = \{T_1, T_2, \ldots, T_n\}$;

6: Selection of trusted fog node to collaborate

$T_F(\Delta t) = \max(T_S)$;

7: **End**;

C. Discussion

In our work, we implemented the process-provenance based trust management scheme. As our proposed trust model is based on the historical interaction of computational tasks, we can calculate the trust value among existing fog devices, not for new arrival fog devices. Because of the lack of past interaction of workload in the new fog device, it will only have the fewest trust value. As our system is intended only for choosing the fog node with the highest trust value to effective collaboration in fog environment, the new arrival fog node in the system may not be selected for collaboration.

We noticed that the storage capacity of the computing workload may impact on the performance of fog node. The data storage for the recording of the computing jobs may be increased time by time.

V. EVALUATION

A. Simulation Setup

To evaluate our proposed process provenance trust mechanism, we illustrate the simulation parameters in this section. We analyze fog infrastructure by using iFogSim [4]. iFogSim is a discrete event simulator for Fog and IoT environments. iFogSim is composed of a set of physical entities such as fog device, sensor, and actuator and logical entities such as AppModule, AppEdge, and Tuple. To simulate the trust value calculation in iFogSim, we used the built-in DCNSFog application.

TABLE I. SIMULATION PARAMETERS

Parameters	Value
Area	4
Camera	4/area
Gateway	1/area
Fog device	7/area
Time window	100 second

The Intelligent Surveillance application has been appraised on several physical infrastructure configurations as shown in Table I. The number of surveillant areas has been varied from 1 to 4. Note that each surveillant area can have a lot of smart cameras that monitoring the area. These cameras are linked to an area gateway that maintains the action in that surveillant area. In the simulated field, each surveillant area has the four number of smart cameras linked to an area gateway that provides Internet access to them.

To simulate the trustworthiness of fog node closely related to the realistic environment is crucial to analyze the effectiveness of our system. Thus, to examine the job failure rate, we adopted log-normal distribution with the parameter value of (0.25, 0.0802) which effects the realistic failure rate of computing jobs [11].

B. Results

We are interested in investigating the effectiveness of our proposed scheme for long-term fog collaboration.

As shown in Fig. 3, we investigated the total trust value of all fog devices with time series to investigate the long-term collaboration. We examine that as the time window increases, the total trust value of all fog nodes in each time window is also increasing. The longer the time window, the number of the historical interaction of computing jobs in each fog node will increase. According to the simulation result, if we start the trust value calculation in 500s, we can get a steady trust value to collaborate. As we can see, the increase in simulation time, our trust result gradually rises. So, we expect that our trust result can effectively be used for collaborating fog nodes.

Fig. 4 plots the average trust value of all fog nodes in each intelligence surveillance area. It can be seen that area 4 has the highest trust value.

However, as each fog device has been configured variedly in terms of resource capacity, the necessary computing resources of each area may different. As area 4 needs only a few of resources than other areas, the processing time of workload of each fog node in this area can be faster. Thus, the average trust value in area4 is high. As a result, we can choose one of the fog devices with the highest trust value for collaboration in the fog environment.

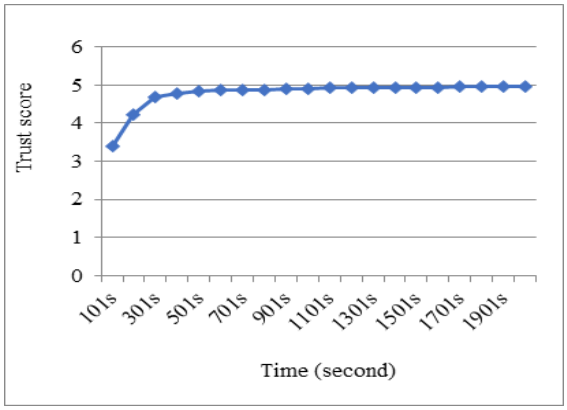

Figure 3. Total trust value with time series

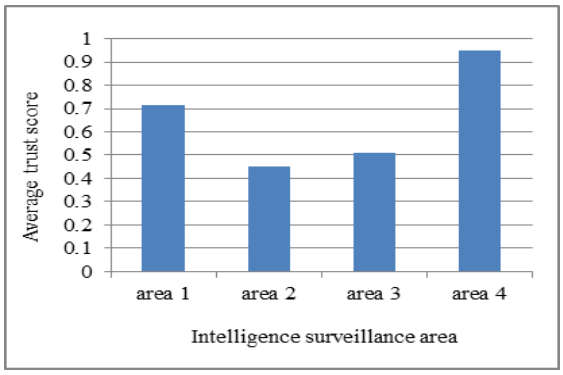

Figure 4. Average trust value with different area

In Fig. 5, the maximum trust score of each fog node in each area is displayed. As our proposed scheme is intended for selecting the fog node with the highest trust value to build the trust model, we analyzed the trust score of each fog node in each intelligence surveillance area.

Although fog devices have a high capacity, some fog devices have a few history interactions of workload. Thus, the completion rate of the workload can high. According to our results, we can determine to select the fog node with id-45 in area4 to collaborate.

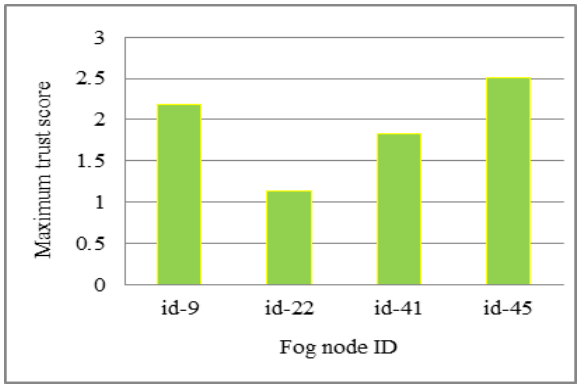

Figure 5. Trust value of fog devices in one area

VI. CONCLUSION

In this paper, we presented the trust evaluation mechanism based on process provenance for fog architecture. We proposed the direct trust mechanism among fog nodes to calculate trust value for fog collaboration. The results are approved through a lot of several analyze and established its effectiveness. We observe the possible expansions of the proposed mechanism in the following indications: we are going to evaluate the completion rate of history interaction of workload in our proposed system with a conventional approach in future work and we plan to consider the data provenance for the data-intensive. In an actual system, massive numbers of IoT devices are allocated. Thus, we will extend to simulate a large number of IoT devices like in the real environment and investigate its effectiveness.

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Software Quality Metrics Calculations for Java Programming Learning Assistant System

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Abstract

A Web-based Java Programming Learning Assistant System (JPLAS) has been proposed to assist Java programming educations in universities. In the code writing problem, the correctness of an answer code from a student is verified by running the test code on JUnit. Besides, their quality should be measured using the metrics to assess them. The currently using plugin could only be measured on eclipse for offline answering in JPLAS. To calculate the metrics and implement in web-based JPLAS, there are several equations that have been reported. In this paper, we find the proper equations to calculate the metrics that provide the same results as from Eclipse plugin. The application results for 45 source codes showed that the adopted metrics equations provide the same results as the plugin.

Keywords: *JPLAS, JUnit, test code, code writing problem*

I. INTRODUCTION

Java is a secure, portable, and platform independent programming language. Java engineers have been demand. Thus, Java is taught in many universities and colleges.

To assist Java educations, *Java programming learning assistant system (JPLAS)* has been developed. In JPLAS, the student can solve the exercise problems both on online and offline using *Eclipse*. JPLAS provides different types of Java programming exercises to cover the different learning levels. The *code writing problem* asks a student to write a whole source code that passes the given *test code* on *JUnit*. Besides, *software metrics* should be measured to assess the quality of the answer code [1].

Software metrics can evaluate the software product under developments, which gives a vision on the quality to make a whole process more successful. These metrics are called *software quality metrics* or *product metrics*. In the code writing problem, seven quality metrics can be measured for assessment of the code. They include NOC - number of classes, NOM – number of methods, CC - cylomatic complexity, NBD – nested block depth, LCOM- the lack of cohesion, MLOC - method lines of code and TLOC – total lines of code [2]. They can be used for the size estimation, the complexity evaluation, and the maintenance of the code.

The currently using plugin can be used on Eclipsed to measure the metrics. Eclipse is used when the students solve the problems on offline. To measure the software metrics of a code, the equations to calculate them must be implementation in the JPLAS server. Unfortunately, there are several different equations to calculate some quality metrics. We need to find proper equation to calculate the metrics that provide the same results from Eclipse plugin.

In this paper, first, equations to calculate each metric are surveyed. Then, the proper one is selected by comparing the results with Eclipse plugin for 45

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answer codes. The results showed that the calculated metrics by the selected equations are equal to those of Eclipse plugin.

II. REVIEW OF JPLAS

In this section, we review JPLAS.

A. JPLAS

In JPLAS, *Ubuntu* is used as the operating system that is running on *VMware*. For web server, *Tomcat* is used to run HTML, JSP and servlets. HTML- hypertext Markup Language is the language to create the webpages. JSP is a script combined with Java code and it is embedded to HTML. Servlet is a small Java program that runs on web server. For database, *MySQL* is used for managing the data.

JPLAS implements supporting functions for teachers and students. In the teacher function, the teacher can show the problems to students by uploading them to the server. In the student function, the student can answer the problems from a Web browser and submit them to the server on online.

Besides, they can answer the problems using Eclipse on offline after receiving the problems by e-mail or downloading the problems from JPLAS, and submit the answer via email.

B. Exercies Problems in JPLAS

In JPLAS, four types of problems are provided. The three problems are the types of fill-in-blank problems to understand the grammar and reading studies. The last problem is for the code writing study. In this paper, the code writing problem is focused for the study of software metrics.

C. Fill-in-blank Problem

In this problem, the students need to fill the correct answers in blanks for a given Java code. The answer is marked by comparing with their original elements in the code. Thus, the original element must be unique correct answer for each blank. The blanks elements are identifiers, variable, reserved words and control symbols.

D. Value Trace Problem

This problem is another type of element fill-in-blank problem that keeps the process of filling and marking the answer. It questions a student about actual values of important variables in the code. The students need to fill the correct values of variables in blanks. In this problem, the output data of variables from the code execution is blanked. The output data may contain one or more values. It blanks the output data line by line.

E. Statement Fill-in-blank Problem

In this problem, the students need to fill the whole statement in a blank for a give Java code. The answer is marked by using the test code on *JUni*t as the code writing problem.

F. Code Writing Problem

In this problem, the students need to write the whole code as the information given in test code. Then, the answer code is tested through the test code on *JUnit* as *test driven development method* (TDD). Besides, their quality is accessed by measuring the seven-quality metrics on *Eclipse plugin*.

G. JUnit

It is an open-source Java framework for unit testing on Java programming language and adopted in JPLAS. It is important in development of TDD - test driven development. Test code is implemented on *JUnit*. Although, test code is a test code programming language, it is rather simple for the Java programmers. This reason is that *JUnit* has been designed for Java programming language. In *JUnit*, one test can be performed by using one method in the *JUnit* library. In the code writing problem, the method whose name starts with 'assert' is used to check the execution results by comparing with the expected method [3].

H. TDD Method

In the TDD method, the following process can be done.

*1)*The source code must be prepared first to write test code from them. Thus, the test code includes the information on model source code, it will be tested in later.

*2)*Then, the answer code is written and tested it on *JUnit* through test code.

*3)*The source code can be re-factored until passing through the test code. Thus, the re-factoring process of a source code becomes easy, because the modified code can be tested instantly.

I. Metric Plugin

Metric plugin for Eclipse is commonly used open source software plugin for metrics calculation. This plugin can measure the various metrics on source code and their results are shown by number in metric view. This plugin is used to measure software metrics to assess the quality of the code for the code writing problem [4][5].

III. Software Metrics

Software metrics are the measurement and prediction of software products, which are essential resources for a project and products relevant for software evolutions. Measurements can be used throughout the software project for quality control by comparing the current measurements with past measurements for similar projects.

A. Overview of Metrics

There are many metrics that can be categorized into process and product metrics. Besides, due to the great interest in the use of object-oriented languages, many object-oriented design metrics has been proposed. *Product* metrics measure size, complexity, quality, and reliability of software product. *Process* metrics measure the various characteristics of the software development process. *Object-oriented* metrics measure the different aspects of object-oriented design, including complexity, cohesion, and coupling. Among them, *product* metrics and *object-oriented* metrics measure the quality of the code.

B. Product Metrics

Product metrics are known as quality metrics. They help improving the quality of the different system components, and comparisons between existing systems. Various kinds of product metrics have been proposed. They include reliability metrics, functionality metrics, performance metrics, usability metrics, cost metrics, size metrics, complexity metrics and style metrics. They are used to measure the properties of the software. Among them, some quality criteria can be used to predict a certain quality of the software [6] and they are as follows:

- NOC - number of classes and DIT - depth of inheritance are measured to assess maintainability and reusability of the program.
- LOC- lines of code is measured to assess the size of the code.
- CC - cyclomatic complexity is measured to assess reliability of the program.

C. Object- Oriented Metrics

Object-oriented designs are more beneficial in software development environment. Object-oriented metrics are used to measure properties of object-oriented designs. The object-oriented metrics measure on class and its design viz; localization, encapsulation, inheritance, polymorphism, and object abstraction techniques, which make the class unique. The object-oriented metrics are defined as follows:

- WMC - Weighted Methods Per Class
- DIT - Depth of Inheritance Tree
- NOC - Number of Children
- CBO - Coupling between Objects
- RFC - Response for a Class
- LCOM - Lack of Cohesion in Methods

IV. Calculation of Seven Metrics

Seven quality metrics are adopted for the code writing problem in JPLAS. They are as the followings:

- NOC - number of classes
- NOM - number of methods
- CC - cyclomatic complexity
- LCOM - lack of cohesion in method
- NBD - nested block depth
- MLOC - method lines of code
- TLOC- total lines of code

The equations of CC and LCOM have many variations, whereas other five metrics have a unique one. In this paper, firstly, all the equations are surveyed. Then, proper equations are selected to calculate them.

A. Number of Classes (NOC)

NOC measures the number of classes within the application package. It is a measure of how many subclasses are going to inherit the methods in the parent class. If a class has many subclasses, it is regarded as the bad design. The lower value of NOC helps maintainability and complexity of codes.

B. Number of Methods (NOM)

NOM measures the number of methods within classes. The number of methods that are local to the class and only those methods can be measured.

C. Cyclomatic Complexity (CC)

CC measures the structural complexity of a procedure by counting the number of independent paths in a method. The paths represent the number of decision points in the code, which include if, while, do-while, for, switch-case-defaults, try-catch finally. The goal of CC is to evaluate the testability and maintainability of a software module [8].

The original complexity is calculated as follows:

$$CC = E - N + 2 \quad (1)$$

Where: CC = cyclomatic complexity
E = the number of edges of the graph
N = the number of nodes of the graph

Then, the improve complexity is defined as the followings [9]:

1) If the source codes contain no decision points, their complexity would be 1 since there is only a single path through the code.

2) If the code has a single IF statement containing a single condition, there would be two paths through the code, one path for TRUE and one path for FALSE.

In above conditions, CC is calculated as follows:

$$CC = E - N + 2P \quad (2)$$

where:
P = the number of connected components

3) An alternate function is used when the cyclomatic complexity is applied to several subprograms at the same time.

$$CC = E - N + P \quad (3)$$

The following example code contains a single IF statement. Thus, it contains the two paths to evaluate the path as TRUE of FALSE.

```
1.  public class Circle{
2.  public static int minFunction(int n1,int n2){
3.              int min;
4.              if(n1>n2)
5.                min=n2;
6.              else
7.                min=n1;
8.              return min;
9.        }
10. }
```

Figure 1. Example code single *If* Statement

Firstly, the statements are transformed into a graph, where every piece of a statement is represented as a node and their flows (sequence of execution of statements) are represented as the edges. For the single program, P is always equal to 1 since it has a single exit point. The cyclomatic complexity may be applied to several subprograms at the same time, where P will be equal to the number of programs. Figure 2 shows the flow chart of the source code containing single IF statements.

In this example, there are seven nodes, seven edges and one connected component. Then, CC= 7-7+2×1=2 is calculated by equation (2).

D. Lack of Cohesion in Methods (LCOM)

LCOM measures the cohesiveness of a class. It represents the difference between two methods whose similarity is zero or not. LCOM can judge the cohesiveness among the class methods. There are several LCOM metrics. The LCOM takes its values in the range 0 to 1.

- If the two methods share at least one field, Q is increased by one. Otherwise, Q is increased by one. It is noted that P and Q are initialized by 0. LCOM is calculated on each pair of metrics as follows [10]:

$$LCOM = (P > Q)? (P - Q):0 \quad (4)$$

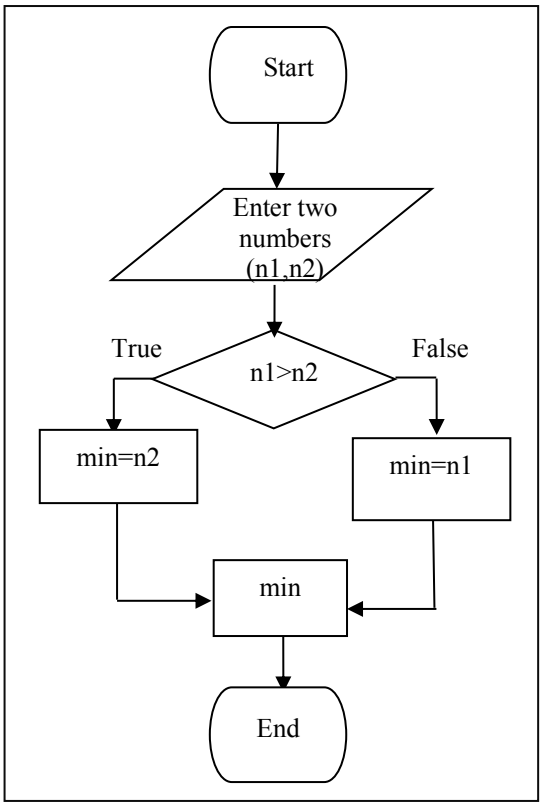

Figure 2. Data flow diagram for source code with single IF statement

- LCOM 1: A low value indicates the high coupling between methods. This also indicates the potentially high reusability and good class design. A high LCOM indicates that a class shall be considered for good design. LCOM = 0 is not a strong evidence that a class enjoys cohesiveness.
- LCOM 2: This is an improved version of LCOM 1.

$$LCOM = 1 - \frac{sum(mA)}{m*a} \quad (5)$$

Where: m = number of methods in class
a = number of attributes in class
mA = number of accessing times of attributes among the methods

- LCOM 3: A completely new expression for cohesion is proposed by Herderson-Sellers, that is called LCOM* [11].

$$\text{LCOM} = \frac{\left(\frac{1}{a}\sum_{j=1}^{a}\mu(Aj)\right) - m}{1 - m} \quad (6)$$

Where: $\mu(Aj)$= number of accessing times of attributes among the methods.

The following example code shows the accessing of attributes (member variables) among the methods. It contains two-member variables: *radius* and *colour*, and three methods: *getRadius*, *getColor* and *getData*. In this example, *radius* is accessed three times by *getRadius* and *getData*. *color* is accessed one time by *getColor*.

By equation (4), LCOM is calculated on the number of accessing times of attributes among the methods. The result is 0 because the value of Q is greater than P where P=0 and Q=4. Then, by equation (5), LCOM is calculated by using the number of methods, attributes and the number of accessing times of attributes. The result is 0.4 where m=3, a=2 and mA=4.

Fig.4 shows the diagram for the methods that accessing the attributes. By equation (6), LCOM is calculated by using the number of methods, attributes and the number of accessing times of attributes. LCOM is 0.5 where m=3, a=2 and $\mu(Aj)$ =4 respectively by equation (6).

According to the equation (4), LCOM is only 0 or 1. According to equation and (5) and (6), LCOM decreases and close to 0, when the accessing times of attributes are more. On the other hand, LCOM increases and close to 1 when the accessing times of attributes are less. The declared variables should be accessed among the methods In this time, equation (6) is selected by the calculation result that is same with plugin in Section V.

```
1. public class Circle{
2.    private double radius;
3.    private String color;
4.    public  Circle (){
5.          radius = 1.0;
6.          color = red;
7.     }
8.     public double getRadius () {
9.          return radius;
10.   }
11.    public String getColor (){
12.         return color;
13.    }
14.    public double getArea (){
15.         return 3.14*radius*radius;
16.    }
17. }
```

Figure 3. Example code for method accessing attributes

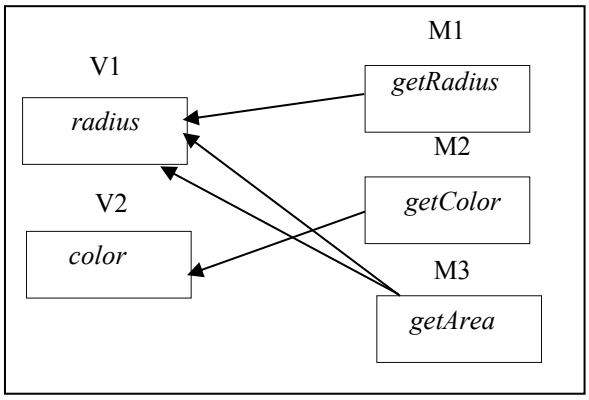

Figure 4. Block diagram of code for methods accessing attributes

E. Nested Block Depth (NBD)

NBD represents the maximum nest depth in a method. The nest depth is given by the number of statements of blocks that are nested due to the use of control structure (branches and loops).

F. Total lines of Code (TLOC)

TLOC measures the total number of lines in the source code. It is calculated by counting on the executable lines, comment and empty lines.

G. Methodl lines of Code (MLOC)

MLOC represents the total number of method lines in method. It is calculated by counting on comments and empty lines.

V. COMPARISON OF METRIC RESULTS

In this section, we evaluate the seven-quality metrics for the 45 source codes for five assignments from nine students in [11]. The first assignments ask the concepts of encapsulation, inheritance and polymorphism. The last two assignments ask the algorithms using those concepts for implementation it.

A. Calculated Metrics

Among the seven metrics, CC and LCOM have several equations to calculate them. For CC, the three equations (1), (2) and (3) are used. For LCOM, equation (6) is used.

B. Comparison of Metrics Values

The metrics values of CC and LCOM for each code are compared between the results by Eclipse plugin and those by the adopted equations to find the proper equations that give the same results. Other values represent the number of classes, methods, branches, and lines of code, which are unique. There is no equation for other metrics values. It can easily be calculated by counting the number of classes, methods, branches and lines of code as described in section IV- A, B, E, F, G. In this paper, the plugin result is represented as *T1* and the equation result is represented as *T2*.

Tables 1-5 show the calculated metrics values of seven metrics for each code. NOC, NOM, NBD, TLOC and MLOC are calculated by counting the number of classes, methods, branches and lines in a code manually. CC is calculated by using Equation (1) (2) and (3). LCOM is calculated by Equation (6) for each code. Then, the values are compared between the plugin and calculated results.

In each assignment, the values NOC, NOM, NBD, TLOC and MLOC are same between T1 and T2. Besides for CC are the same values between them. The values of LCOM are slightly varied between T1 and T2 because of the difference of significant digits. Thus, the adopted equations and the counted number for them are the same as those in Eclipse plugin. In assignment 1, 2 and 3, they don't need the conditions or branches (loops) to implement the concepts of OOP: encapsulation, inheritance and polymorphism. Thus, their CC and NBD are always 1 by both T1 and T2. In assignment 4 and 5, it needs the conditions or branches (loops) to implement the algorithms using the OOP concepts. In assignment 4, its CCs are 2-3 and NBD is 1-3. In assignment 4, its CCs are 2-4 and NBD is 1-3. The values are varied among the codes depending on the student's implementation on code.

TABLE I. COMPARISON OF METRIC VALUES FOR ASSIGNMENT 1

	Assignment 1							
	NOC		*NOM*		*MLOC*		*TLOC*	
	T1	*T2*	*T1*	*T2*	*T1*	*T2*	*T1*	*T2*
S1	1	1	6	6	8	8	26	26
S2	1	1	6	6	8	8	26	26
S3	1	1	6	6	8	8	26	26
S4	1	1	6	6	8	8	26	26
S5	1	1	6	6	8	8	27	27
S6	1	1	6	6	8	8	26	26
S7	1	1	6	6	8	8	26	26
S8	1	1	6	6	8	8	26	26
S9	1	1	6	6	8	8	26	26
	NBD		*CC*		*LCOM*			
	T1	*T2*	*T1*	*T2*	*T1*	*T2*		
S1	1	1	1	1	0.5	0.5		
S2	1	1	1	1	0.5	0.533		
S3	1	1	1	1	0.5	0.533		
S4	1	1	1	1	0.5	0.533		
S5	1	1	1	1	0.7	0.667		
S6	1	1	1	1	0.5	0.533		
S7	1	1	1	1	0.7	0.667		
S8	1	1	1	1	0.6	0.6		
S9	1	1	1	1	0.7	0.667		

TABLE II. COMPARISON OF METRIC VALUES FOR ASSIGNMENT 2

	Assignment 2							
	NOC		*NOM*		*MLOC*		*TLOC*	
	T1	*T2*	*T1*	*T2*	*T1*	*T2*	*T1*	*T2*
S1	2	2	5	5	5	5	28	28
S2	2	2	5	5	5	5	24	24
S3	2	2	5	5	5	5	24	24
S4	2	2	5	5	5	5	24	24
S5	2	2	12	12	15	15	48	48
S6	2	2	6	6	9	9	28	28
S7	2	2	5	5	5	5	24	24
S8	2	2	5	5	5	5	22	22
S9	2	2	9	9	11	11	40	40
	NBD		*CC*		*LCOM*			
	T1	*T2*	*T1*	*T2*	*T1*	*T2*		
S1	1	1	1	1	0.5	0.5		
S2	1	1	1	1	0.5	0.5		
S3	1	1	1	1	0.5	0.5		
S4	1	1	1	1	0.5	0.5		
S5	1	1	1	1	0.7	0.68		
S6	1	1	1	1	0.5	0.5		
S7	1	1	1	1	0.7	0.667		
S8	1	1	1	1	0.5	0.5		
S9	1	1	1	1	0.7	0.667		

TABLE III. COMPARISON OF METRIC VALUES FOR ASSIGNMENT 3

	Assignment 3							
	NOC		*NOM*		*MLOC*		*TLOC*	
	T1	*T2*	*T1*	*T2*	*T1*	*T2*	*T1*	*T2*
S1	1	1	3	3	3	3	14	14
S2	1	1	3	3	3	3	12	12
S3	1	1	3	3	3	3	12	12
S4	1	1	3	3	3	3	12	12
S5	1	1	3	3	9	9	21	21
S6	1	1	3	3	3	3	12	12
S7	1	1	3	3	3	3	12	12
S8	1	1	3	3	6	6	15	15

S9	1	1	3	3	3	3	12	12
	NBD		***CC***		***LCOM***	
	T1	***T2***	***T1***	***T2***	***T1***	***T2***
S1	1	1	1	1	0	0
S2	1	1	1	1	0	0
S3	1	1	1	1	0	0
S4	1	1	1	1	0	0
S5	1	1	1	1	0	0
S6	1	1	1	1	0	0
S7	1	1	1	1	0	0
S8	1	1	1	1	0	0
S9	1	1	1	1	0	0

TABLE IV. COMPARISON OF METRIC VALUES FOR ASSIGNMENT 4

	Assignment 4							
	NOC		***NOM***		***MLOC***		***TLOC***	
	T1	***T2***	***T1***	***T2***	***T1***	***T2***	***T1***	***T2***
S1	1	1	5	5	29	29	50	50
S2	1	1	4	4	37	37	53	53
S3	1	1	4	4	37	37	53	53
S4	1	1	3	3	7	7	19	19
S5	1	1	4	4	20	20	36	36
S6	2	2	9	9	26	26	55	55
S7	1	1	5	5	26	26	44	44
S8	1	1	4	4	24	24	40	40
S9	1	1	5	5	39	39	60	60
	NBD		***CC***		***LCOM***	
	T1	***T2***	***T1***	***T2***	***T1***	***T2***
S1	3	3	4	4	0.4	0.4
S2	3	3	3	3	0.6	0.667
S3	3	3	3	3	0.6	0.667
S4	1	1	2	2	0.5	0.5
S5	2	2	2	2	0.8	0.8
S6	2	2	2	2	0.45	0.45
S7	2	2	2	2	0.2	0.2
S8	2	2	3	3	0.6	0.6
S9	3	3	3	3	0.5	0.5

TABLE V. COMPARISON OF METRIC VALUES FOR ASSIGNMENT 5

	Assignment 5							
	NOC		***NOM***		***TLOC***		***MLOC***	
	T1	***T2***	***T1***	***T2***	***T1***	***T2***	***T1***	***T2***
S1	3	3	5	5	10	10	34	34
S2	2	2	5	5	28	28	49	49
S3	2	2	5	5	28	28	49	49
S4	2	2	4	4	9	9	26	26
S5	2	2	5	5	17	17	39	39
S6	2	2	9	9	26	26	54	54
S7	2	2	6	6	36	36	59	59
S8	2	2	5	5	33	33	55	55
S9	2	2	13	13	45	45	87	87
	NBD		***CC***		***LCOM***	
	T1	***T2***	***T1***	***T2***	***T1***	***T2***
S1	2	2	2	2	0.5	0.5
S2	2	2	2	2	0.8	0.8
S3	2	2	2	2	0.8	0.8
S4	1	1	2	2	0.5	0.5
S5	2	2	2	2	0.9	0.9
S6	2	2	2	2	0	0
S7	2	2	3	3	0.1	0.1
S8	2	2	3	3	0.6	0.6
S9	3	3	2	2	0.5	0.5

VI. CONCLUSIONS

In this paper, we surveyed equations for calculating software quality metrics, verified through comparison between T1 and T2 and found the equations that provide the same values as the Eclipse plugin through applications to 45 source codes. In future works, we will implement the equations in JPLAS and evaluate the quality of source codes from students on real time.

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