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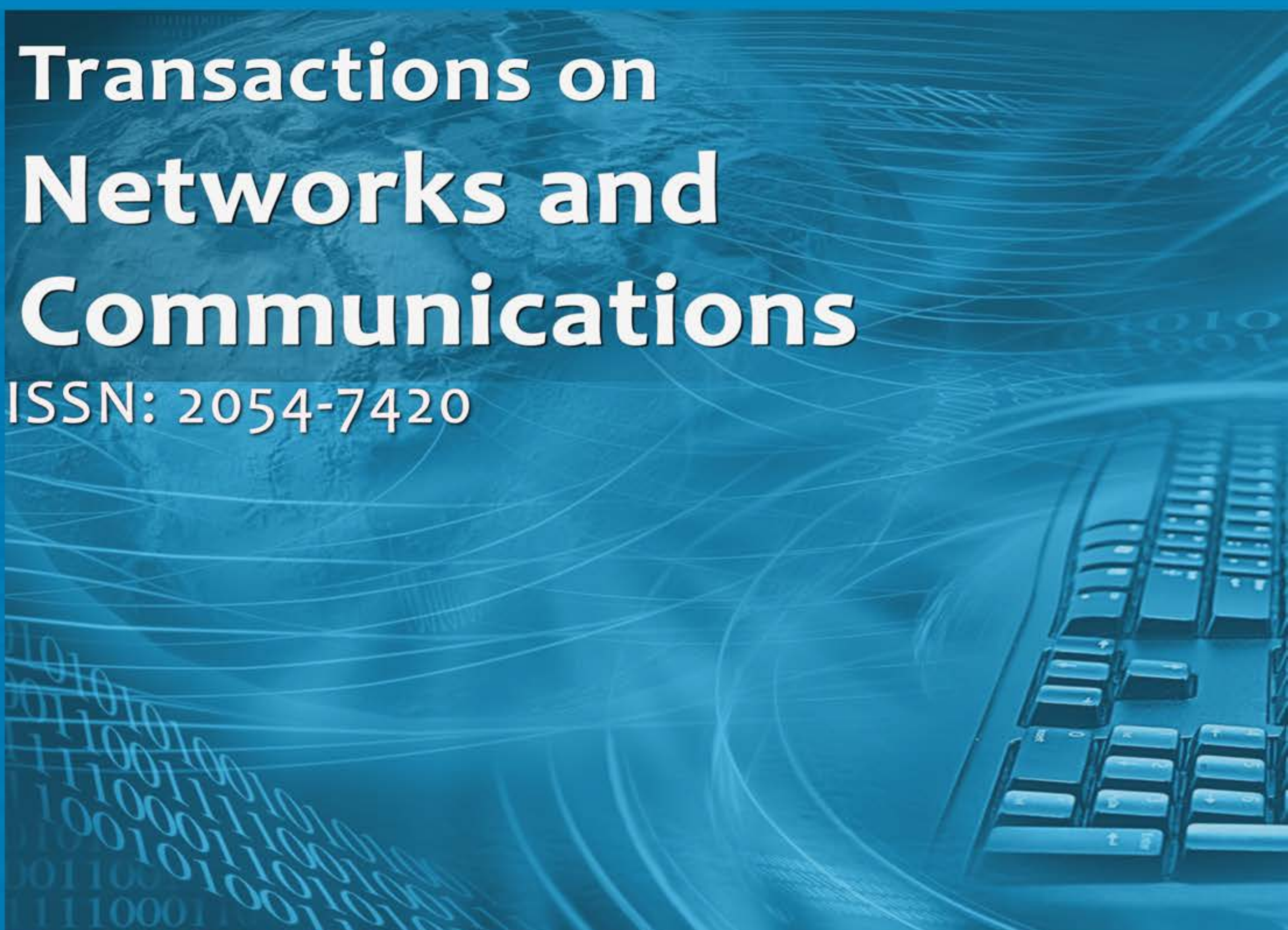

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Designing Networks with Low Structural Congestion via Game Theory and Linear Programming

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ABSTRACT

We propose a network topology design approach that targets the reduction of structural congestion in a directed acyclic network. What we mean by structural congestion is that a node has much higher in-degree than out-degree in a directed network. We approach the issue using a network design game model. In this model we consider multiple sources and one destination. Each node is willing to connect to other nodes but it should pay the price of whole paths it uses to send traffic to the destination. The model yields a weight for each link. We show that if these weights are used to compute shortest paths, then a network topology is obtained with a low level of structural congestion.

The proposed method has two phases. In Phase I, we solve a linear optimization problem in order to find the optimum link weights. In Phase II, each node optimizes its own individual objective function, which is based on the weights computed in Phase I. We show that there exists a Nash Equilibrium which is also the global optimum. In order to measure the penalty incurred by the selfish behavior of nodes, we use the concept called price of anarchy. Our results show that the price of anarchy is zero.

Keywords— Communication network; Game theory; Linear programming

1 Introduction

Communication network design methods and algorithms are approached with various types of design goals. Minimum vulnerability, fault tolerance and quality of services are often used in this context [1].

As network nodes become more intelligent, distributed algorithms become increasingly dominant. Although a centralized algorithm which optimizes the entire network configuration would maximize efficiency and utilization, it is not as stable as distributed algorithm. Stability in a network means that if some nodes fail, other nodes have the capability to reconfigure themselves and recover from the failure. This idea can lead to a decision making algorithm that is executed in each node separately to optimize the global benefit.

One step further in this direction is when a node does not know the global benefit or does not care about it. In this situation a network involves selfish agents, making decisions to optimize their own benefit [2]. Social and biological networks are examples of such selfishly behaving agents that form a network. Game theory is a useful tool to analyze and predict the behavior of this kind of networks.

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In this work, we study a directed acyclic network design game in the light of structural congestion consideration. Each node in a network which has a high in-degree is a bottleneck. It is desirable to avoid such a structural bottleneck, as it can easily lead to traffic congestion.

Our main objective here is to show that there exists a well-defined utility function in which the selfish behavior of each node leads to a network topology with minimum structural congestion. To do that first we convert a minimum structural congestion problem into a shortest path routing problem, in which link weights are obtained as the output of a linear optimization task. Then we construct a utility function in order to encourage each node to use paths with minimum overlap. The path set forms a new network which has a minimal structural congestion.

The rest of this paper is organized as follows. After discussing related work in section II, we define the concept of structural congestion and optimization framework for analyzing network topology in section III. In section IV we derive a condition in which selfish behavior of each node can lead to an optimum. Finally, conclusion is presented in section V.

2 Related Work

The design of various networks have been studied in sociology, natural sciences and engineering for a long time [3]. Optimization and graph theory was the most useful tool in this field, since Myerson introduced a new network design model using game theory for social and economic networks [4]. After that, the concept of game theoretic models have been used in different communication networking contexts, such as routing [5], flow control [6] and dynamic access control in wireless networks [7].

Nash Equilibrium has been considered as a way to quantify the performance associated with selfish behavior of each player. Such equilibria are inefficient [12]. The lack of global control can lead to suboptimal network performance. The "price of anarchy" is a concept in game theory which measures the inefficiency of a system due to selfish behavior of each player [13].

A comprehensive study of game theory based communication network design is [1], which involves three important design considerations, namely the price of establishing a link, path delay, and path proneness to congestion. They showed that there exists an equilibrium point which is a global optimum.

The cost function which they considered in [1] for each player in a network design game considering path congestion is:

$$C(v_i) = \max_{v_k \in V} \max_{v_j \in l(v_i, v_k)} \eta_G^{in}(v_k) \tag{1}$$

In which $\eta_G^{in}(v_k)$ is the input degree of a node v_k in a graph G, and $l(v_i, v_k)$ denotes the path connecting v_i and v_k. In this method each node is required to connect to all other nodes and they show that a directed ring is both an optimum and equilibrium.

In this study, we focus on the *structural congestion* of the network. For our purposes, the network can be represented by a weighted directed acyclic graph.

3 Structural Congestion

A path in a network is a sequence of links, each link (except the first) having the same start node as the end node of the previous link in the sequence. Each link has a utilization factor, which we call

Link Utilization (LU). If we view the network topology as a set of paths from a source to a destination, it contains several links which have different LU. A path's proneness to congestion is depending on the maximum LU on the path from a source to a destination. Let us look at a node v_i in the network, which is described using the graph $G(N,E)$. Let η_i^{in} and η_i^{out} be the input and output degree of v_i. We define the Degree Ratio (DR) for each node as follow:

Definition: The degree ratio of a node $i \in N$ is $DR_i = \dfrac{\eta_i^{in}}{\eta_i^{out}}$.

Assuming all links have unit capacity, the quantity DR_i shows the structural congestion at the node. High DR_i means node $i \in N$ is a bottleneck and can be a point of congestion. There is a direct relationship between DR_i in a network and the Maximum Link utilization (MLU) which is described in the following conjecture.

Conjecture: A set of paths in a directed acyclic network which minimizes MLU, will form a new network which minimizes (at least approximately) DR_i for all $i \in N$ and carries the same amount of traffic.

Minimizing MLU means finding a set of paths between a source and a destination, such that these paths split the input traffic as much as possible and, at the same time, have a minimum overlap. First we analyze the problem of minimizing MLU, because it is a linear optimization problem. Consider a directed acyclic graph $G(N,E)$ which represents the network. $C_{ij}:(i,j)\in E$ is a set of edge capacities and (s_k,t_k) is a set of source-destination pairs for each session $k \in K$. The percentage of traffic on a link $(i,j)\in E$ that belongs to session k is X_{ij}^k. With these notations, the formulation is [8]:

$$\begin{aligned}&\min_X MLU\\ &\sum_{j:(i,j)\in E} X_{ij}^k - \sum_{j:(j,i)\in E} X_{ji}^k = \begin{cases} 1 & i\varepsilon S\\ -1 & i\varepsilon T\\ 0 & otherwise\end{cases}\\ &\sum_{k\in K} X_{ij}^k \le C_{ij}.MLU\\ &X_{ij} \ge 0 \in \varepsilon\end{aligned} \tag{2}$$

Using duality theory we can write the dual optimization problem as follows:

$$\begin{aligned}&\max_{P,W} \sum_{k\in K} p_{t_k}^k\\ &\sum_{(i,j)\in E} C_{ij} w_{ij} = 1\\ &p_i^k - p_j^k \le w_{ij}\\ &p_{s_k}^k\\ &w_{ij} \ge 0\end{aligned} \tag{3}$$

Because the primal and dual problems are both linear, strong duality holds and according to complementary slackness in the KKT theorem if $\hat{X}_{ij}^k$ is an optimal solution for the primal problem, and $\{\hat{w}_{ij}, \hat{p}_{ij}^k\}$ is an optimal solution for the dual, then we have:

$$\hat{X}_{ij}^{k}.(\hat{p}_i^k - \hat{p}_j^k + \hat{w}_{ij}) = 0 \tag{4}$$

This equation indicates that if session k uses link $(i, j) \in E$ then $p_j^k - p_i^k = w_{ij}$. According to the Duality Theorem, if $\{\hat{w}_{ij}\}_{(i,j)\in E}$ is used as link metric in a shortest path algorithm, all non-empty links $(X_{ij}^k > 0)$ will be selected in a shortest path algorithm procedure [8]. As a result if any shortest path algorithm uses $\{\hat{w}_{ij}\}_{(i,j)\in E}$ as link weights we will have set of paths between a source and destination which has an important character. The path set splits the input traffic as much as possible through the network and at the same time has minimum number of overlap links.

A network topology with minimum structural congestion means that DR_i is close to one. Let us consider a weighted directed acyclic graph which represents a network with only one source-destination pair and the capacity of all links are 1. Weights are calculated on the basis of the dual optimization problem discussed above. If we run any shortest path algorithm over such a network we obtain a set of paths Ω. If we delete any link $(i, j) \in E$ which is not on a member of Ω we will have a new weighted acyclic graph which represents a new network. All nodes in the new network have an equal or smaller DR_i than the old one.

4 Network Design Model

In this section we study the performance of a non-cooperative network. This means, each node (player) tries to maximize its own benefit. The network design goal is minimizing the structural congestion. Node v_i gains α_i by connecting to any node in the network. So each node tries to make a connection to as many nodes as possible. By connecting to each node it must calculate the length of a path from itself to a destination. The gain a node can achieve by connecting to others minus the summed length of all paths heading to destination form the utility function of each node as follows.

Node Utilization:

$$u_G(v_i) = \alpha_i |S_i| - \sum_{p\in P} d^{\,p}_{G(W)}(v_i, v_t) \tag{5}$$

Which $|S_i|$ is the number of output links in node v_i and $d^{\,p}_{G(W)}(v_i, v_t)$ is a distance between node v_i and the destination using path $p \in P$ in the designed network using links weight W.

The network utilization is the sum of all node utilization functions.

Network Utilization:

$$U_G = \sum_{i\in N} u_G(v_i) \tag{6}$$

Optimum solution for such a game happens when we have a maximum $U_G = \sum_{i\in N} u_G(v_i)$. But in order to find equilibrium point we need to analyze the selfish behavior of each node. For that purpose consider Figure 1 as a part of a network. Node v_i is deciding to stay on its current strategy (connection to other nodes) or deviate (drop a connection or make a new one) based on the maximum utilization function.

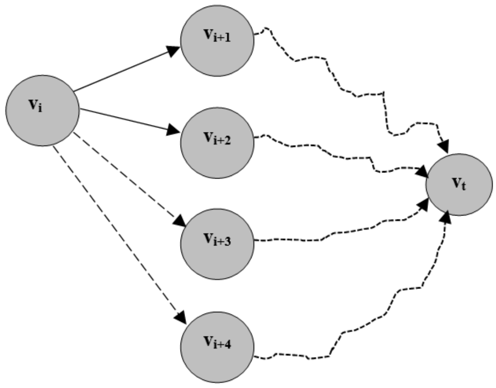

Figure 1: Node V_i decision strategies

Consider node v_i in the network. It is already connected to nodes v_{i+1} and v_{i+2}. It should decide to connect to nodes v_{i+3} and v_{i+4} or not. The current topology is represented by the graph G_1, if it connect to v_{i+2} the graph will be G_2 and if it connect to both v_{i+1} and v_{i+2} the graph will be G_3. Based on the weight system in the network the distance from nodes v_{i+1}, v_{i+2}, v_{i+3} and v_{i+4} to destination are l_{i+1}, l_{i+2}, l_{i+3} and l_{i+4} respectively. The utility of node v_i is:

$$u_{G_1}(v_i) = 2\alpha_i - (w_{i,i+1} + l_{i+1} + w_{i,i+2} + l_{i+2}) \tag{7}$$

$$u_{G2}(v_i) = 3\alpha_i - (w_{i,i+1} + l_{i+1} + w_{i,i+2} + l_{i+2} + w_{i,i+3} + l_{i+3}) \tag{8}$$

$$\begin{aligned} u_{G3}(v_i) = 4\alpha_i - (w_{i,i+1} + l_{i+1} + w_{i,i+2} + l_{i+2} + w_{i,i+3} + l_{i+3} \\ + w_{i,i+4} + l_{i+4}) \end{aligned} \tag{9}$$

Suppose that based on the weight system, links $(i,i+1)$, $(i,i+2)$ and $(i,i+3)$ are on the shortest paths. So we have:

$$l^i_{Sh-P} = w_{i,i+1} + l_{i+1} = w_{i,i+2} + l_{i+2} = w_{i,i+3} + l_{i+3} \tag{10}$$

If we want that the selfish behavior of the node v_i leads to optimum topology, then the following conditions must hold:

$$u_{G_1}(v_i) < u_{G2}(v_i) \tag{11}$$

$$u_{G2}(v_i) > u_{G3}(v_i) \tag{12}$$

So we have:

$$w_{i,i+1} + l_{i+1} < \alpha_i < w_{i,i+4} + l_{i+4} \tag{13}$$

If we consider λ^i_{Sh-P} as the length of a second shortest path from the node v_i to the destination we have:

$$l^{i}_{Sh-P} < \alpha_i < \lambda^{i}_{Sh-P} \tag{14}$$

This is the condition in which selfish behavior of each node in the network will lead to optimum topology with minimum structural congestion. Now the question is if there is any upper and lower bound for α in general. Using topological sorting theorem [9] we can find such a bound. Based on topological sorting theorem a directed acyclic graph can be represented in way that nodes index increase when they get closer to the destination and there is no link (m,n) if $m > n$. For example a directed acyclic graph with 4 nodes after topological sorting is shown in figure 2.

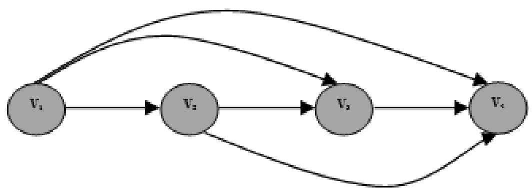

Figure 2: Topological Sorting

After topological sorting we suppose that node 1 is the source and node N is the destination. Now it is clear that after using weight set which is the solution of dual optimization problem in section III we have $l^{i}_{Sh-P} > l^{i+1}_{Sh-P}$. So the lower bound for α is l^{1}_{Sh-P} which is the shortest path from source to the destination. Also we have $\lambda^{i}_{Sh-P} > \lambda^{i+1}_{Sh-P}$. So the upper bound for α is:

$$\lambda^{N-2}_{Sh-P} = \max\{w_{N-2,N-1} + w_{N-1,N}, w_{N-2,N}\} \tag{15}$$

So we have:

$$l^{1}_{Sh-P} < \alpha < \lambda^{N-2}_{Sh-P} \tag{16}$$

Now consider the network in figure 2. The question is what is the upper and lower bound for α in this network. Table I shows the optimal weights which calculate using dual optimization problem in section III.

Table 1: Optimum Weights

$p_1 - p_2 = w_{12}$	1
$p_1 - p_3 = w_{13}$	1
$p_1 - p_4 = w_{14}$	2
$p_2 - p_3 = w_{23}$	3
$p_2 - p_4 = w_{24}$	1
$p_3 - p_4 = w_{34}$	1

In this case upper and lower bound is:

$$l^{1}_{Sh-P} = 2$$

$$\lambda^{N-2}_{Sh-P} = 4$$

So $\alpha = 3$ satisfies the condition. After applying $\alpha = 3$ in the node utility function, node v_2 can improve its utility function by deviate from current strategy to the one which has no connection to node v_3. As a result we have network with better structural congestion. Applying this method to all nodes the result would be a network topology with minimum structural congestion.

In order to analyze the price of selfish behavior there is two important concepts which are price of stability and price of anarchy. The price of stability is the ratio between best objective function value in equilibrium point and the optimum network utilization function. On the other hand price of anarchy is the ratio between worse objective function value in the equilibrium and the optimum network utilization function [10]. In this section we showed that price of stability is one and anarchy is free if each node applies the node utilization function. Otherwise price of anarchy is depend on α and $\{w_{ij}\}_{(i,j)\in E}$ [1].

Figure 3 shows how the optimization method provides inputs for our network design game.

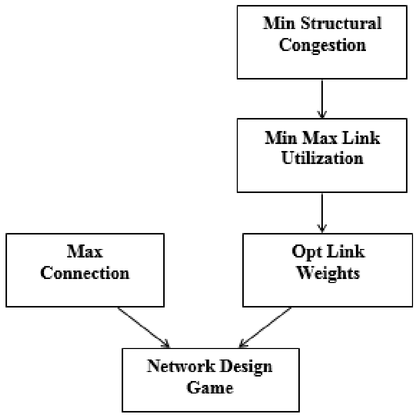

Figure 3: Algorithm Flowchart

It is worth mentioning that the described method can be implemented in a network using distributed algorithms like the Bellman-Ford Algorithm [11]. It means that it is not necessary for each node to have information about the whole network. It is only needed to know the parameter α , the weights of its outgoing links and the distance of its neighbors to the destination. Having this information is sufficient to find an optimum strategy.

5 Simulation

For the simulation we consider a directed acyclic network with 20 nodes. All links have a capacity one and we consider node 1, 2 and 3 as a source of traffic and node 20 as the destination. Figure 4 shows the network topology. Maximum degree ratio is 19 in this network. Each node minimizes its own objective function based on optimum link weights and its desire to make more connection.

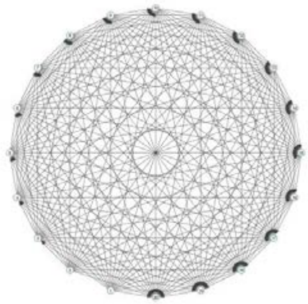

Figure 4: Network Topology

After solving the dual optimization problem we have $l^{1}_{Sh-P} = 4.3$ and $\lambda^{18}_{Sh-P} = 14.5$. Figure 4 shows that no structural congestion is a result of choosing $4.3 < \alpha < 14.5$, it means that $DR_i = 1$ for all $i \in N$. As α deviates from the constraint each node is more willing to make a new connection and it leads to more structural congestion. For example if we choose $\alpha = 20$ degree ratio of nodes 16 and 17 are high and they can be considered as a network bottleneck.

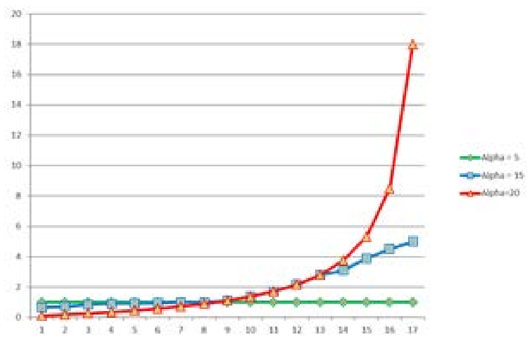

Figure 5: Degree ratio for each node using different alpha

6 Conclusion

This paper investigates the question "*how non-cooperative nodes in a network can create an efficient network?*" We have studied the result of the selfish behavior of nodes, and compares it to the situation in which there is a central control unit in the network. Central control can force all nodes to use a predefined strategy in which the network utilization is optimum.

Based on the discussion in section IV if we fix the benefit of establishing a new link for each node, α ,in a way that satisfies the condition $l^{1}_{Sh-P} < \alpha < \lambda^{N-2}_{Sh-P}$, the price of stability will be one and also the price of anarchy will be zero in this network design game.

REFERENCES

[1] Nahir, A., et al. Topology Design of Communication Networks: A Game-Theoretic Perspective. IEEE/ACM Trans. Netw. 22(2): (2014) 405-414.

[2] Page Jr, F. H., et al. Strategic basins of attraction, the path dominance core, and network formation games. Games and Economic Behavior 66.1 (2009): 462-487.

[3] Jackson, M., et al. On the formation of interaction networks in social coordination games. Games and Economic Behavior 41.2 (2002): 265-291.

[4] Aumann, R., et al. Endogenous formation of links between players and coalitions: an application of the Shapley value. The Shapley Value (1988): 175-191.

[5] Fotakis, D., et al. The structure and complexity of Nash equilibria for a selfish routing game. Theoretical Computer Science 410.36 (2009): 3305-3326.

[6] Fang, Z., et al. Fair bandwidth sharing algorithms based on game theory frameworks for wireless ad-hoc networks. INFOCOM 2004. Twenty-third AnnualJoint Conference of the IEEE Computer and Communications Societies. Vol. 2. IEEE, 2004.

[7] Ji, Z., et al. Cognitive radios for dynamic spectrum access-dynamic spectrum sharing: A game theoretical overview. Communications Magazine, IEEE 45.5 (2007): 88-94.

[8] Wang, Y., et al. Internet traffic engineering without full mesh overlaying. INFOCOM 2001. Twentieth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE. Vol. 1. IEEE, 2001.

[9] Varol, Y., et al. An algorithm to generate all topological sorting arrangements. The Computer Journal 24.1 (1981): 83-84.

[10] Roughgarden, T. Selfish routing and the price of anarchy. Vol. 174. Cambridge: MIT press, 2005.

[11] Awerbuch, B., et al. Approximate distributed bellman-ford algorithms. Communications, IEEE Transactions on 42.8 (1994): 2515-2517.

[12] G. Debreu, A social equilibrium existence theorem, Proc. Nat. Acad. Sci., vol. 38, pp. 886–893, Oct. 1952.

[13] Koutsoupias, E., et al. Worst-case equilibria. STACS 99. Springer Berlin Heidelberg, 1999.

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Online Road Traffic Accident Monitoring System for Nigeria

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ABSTRACT

The collection, management and dissemination of road traffic accident (RTA) related events has posed a serious problem to Nigerian road safety officers and all stakeholders alike. This is especially true because of the absence of a central repository from which all accident related information can be stored and managed. Where available, these information are available in paper based form and this poses a bottleneck in updating the available information. In this study, a Web based Road Traffic Accident Monitoring System (RTAMS) was developed for Nigerian.

The Road Traffic Accident Monitoring System (RTAMS) was developed using Adobe Dreamweaver and Notepad++ as the Integrated Development Environments, HTML, CSS and JavaScript were used for the frontend, PHP was used as the scripting language, and MySQL served as the database server. Most of the languages and tools used were open source which ensured that the application would be robust, reusable, cheap and highly scalable.

The result of the developed system shows that road safety officers, users, policy makers and all other stakeholders can be able to register, login, submit reports and run queries on information that has been previously entered into the system such as the accidents that occurred on a particular route or the accident in which an identified victim was involved. Policy makers can run these queries in order to take appropriate steps in minimizing road traffic accident occurrences.

In conclusion, this system will help create a paperless alternative to the present method of RTA information and thus make information dissemination quicker and also improve first aid response to accident occurrences.

Keywords: Road Accident, Traffic, Monitoring System, Vehicle

1 Introduction

Road traffic Accidents (RTA) is one of the major causes of mortality and morbidity around the world and it has been discovered that low and middle income countries are the most affected (Krug et al, 2000). The World Health Organization estimates that more than 3,000 people are killed every day in road traffic accidents globally, with at least 30,000 others injured or disabled. This adds up to over 1 million people killed and between 20 – 50 million injured or crippled in road traffic accidents each year (Krug et al, 2000). The increased rate of fatal road accidents worldwide has been attributed to increased motorization and explosion (WHO, 1984, Atubi, 2010); statistics indicate that over 90 percent of traffic accident situations in Nigeria can be attributed to driver errors (Aworemi et al, 2009).

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Motor vehicle crashes have also been discovered to be the leading cause of death in adolescents and young adults (Taket, 1986) and of the estimated 856,000 road deaths occurring annually worldwide, 74% are in developing countries (World Bank, 1990). While the proportion and absolute number of traffic fatalities have reduced by more than 20% in industrialized nations, a number of developing countries have experienced dramatic increases (Ross et al, 1991, Akinyemi, 2009). Nigeria and Kenya for instance have experienced a fivefold increase in traffic related fatalities over a period of 30 years of observation. Also it has been discovered that Asian and African countries, with relatively low vehicle densities, are experiencing substantially higher fatality rates per 10,000 vehicles than the industrialized European and North American Nations (WHO, 1984; Atubi, 2010).

The global costs of road traffic injuries are enormous, one report estimates the global costs of road crashes is about $518 billion annually in US Dollars, and ranges in percentage of GNP (Gross National Product) from 0.3% in Vietnam to almost 5% of GNP in the USA and Malawi (Jacobs et al, 2000). In another report it is stated that traffic crashes impact the economy of developing countries at an estimated cost of 1 – 2 % of a country's GNP per annum, as a result of mortality, morbidity, and property – related costs (WHO, 1989; Akinyemi, 2009).

Causes of motor vehicle crashes involve the interaction of multiple factors that include people, vehicles and the road environment. Human error is estimated to account for between 64% and 95% of all causes of traffic crashes in developing countries (Haddon, 1980; TRL, 1990, Atubi, 2010). A large number of old vehicles often carry more passengers than they are designed to carry and a lot of them lack safety belts and helmets; apart from these factors, poor road design and maintenance is a factor that contributes to the high rate of crashes in developing countries.

The major causes of road traffic accidents could be classified under three broad subheadings: vehicle-related factors, human-related factors, and environment related factors (Gungul, 2012). Vehicle – related factors include vehicle design, the vehicle body, the brake system, the vehicle tyres, the vehicle lights and the engine. Every vehicle is designed for a specific maximum load in all areas so it is no surprise that when it is subjected to stress above the provisions of the design specifications, accelerated wear and tear set into the vehicles (Gungul, 2012). The brake subsystem, working jointly with the accelerator synchronizes the speed of vehicles; any malfunctioning of the brake sub-system is to be taken as a potential source of accident. Also, when tyres are overinflated, thoroughly worn out or when there is failed indicator lights or non-existent headlights, accidents could occur. The sudden failure of the engine sub-system (which may be considered as the "brain" of the vehicle), if mismanaged could cause an accident even if an experience driver is at the helm (Gungul, 2012).

As regards Human-related factors, studies have clearly shown that the single most important contributing factor to road traffic accidents in Nigeria is the attitude of the driver to driving code and etiquette (Aworemi et al, 2009). Human-related issues include fatigue and sleepiness, faulty preparation, ignorance of highway codes or traffic orders, driving under the influence of drugs and/or alcohol, and inexperience.

Environmental factors contribute greatly to the rate of road accidents in Nigeria today and some of the well –known factors are fog, sunrays, mist and rain. Injuries especially road traffic injuries (RTIs) are linked to the environmental factors, also a significant number of vehicular accidents can be traced to the condition of the road. Recent studies have shown that the road is another major factor in road accidents in Nigeria (Asolor et al, 2008). Deficiencies of Nigerian roads are due largely to inadequate road design specification and maintenance (Akinyemi, 2009). Other significant factors include the frequency of potholes on the roads, the indiscriminate location of police check points and the reluctance of the appropriate authorities to continually improve on the condition of the roads.

Engineering interventions are believed to be the most effective interventions for road traffic accidents (Peden et al, 2004). Improvement in road design, visibility, speed changes, are some such interventions, though their implementations requires information on areas with higher concentration of RTAs.

The aim of this paper is to develop a web-based road traffic accident monitoring system to replace the existing paper – based one. This monitoring system can be used by road safety officers to respond quickly to road accidents, to survey and record such occurrences, for other road traffic stakeholders to view records and for decision making bodies to make policies to reduce road traffic accidents.

2 Road Traffic Accidents in Nigeria

Statistics indicate that over 90 percent of traffic accident situations in Nigeria can be attributed to driver errors (Aworemi et al, 2009). Road accidents appear to occur regularly at some flash points such as where there are sharp bends, potholes and at bad sections of the highways. At such points over speeding drivers usually find it difficult to control their vehicles, which then result to fatal traffic accidents, especially at night (Atubi, 2009).

Cases of fatal road traffic accidents are reported almost daily on the major highways in Lagos State. Various categories of vehicular traffic are also involved in these fatal road traffic accidents in the state. Research in this area have focused on cases of road traffic accidents, collation of road traffic accident statistics and impact assessment of road safety campaign (Becker, 1996; Gozias et al, 1997 and Odero et al, 2003).

At the local level research in this area are concentrated on the effects of land use and human factors on road traffic accidents (Onokala, 1995). Motor vehicle crashes are the leading cause of death in adolescents and young adults (Taket 1986; Atubi and Onokala, 2009) and of the estimated 856,000 road deaths occurring annually worldwide, 74% are in developing countries (Atubi, 2000). Dramatic increases in the proportion and absolute number of traffic facilities have been witnessed in a number of developing countries, while they decreased by more than 20% in industrialised nations (Ross et al, 1991). In Nigeria (Oluwasanmi, 1993; Atubi, 2009b, 2009e and 2010c), a fivefold increase in traffic related fatalities was observed over the last 30 years.

African and Asian countries, with relatively low vehicle densities, are experiencing substantially higher fatality rates per 10,000 vehicles than the industrialised European and North American States (WHO, 1984; Atubi and Onokala, 2009).

In Nigeria, road traffic accident situation over the last three decades has been particularly disturbing. In 1976, there were 53,897 road traffic accidents resulting in 7,717 deaths. Although in 1981, the magnitude reduced to 5,114 accidents, but the fatality increased to 10,236 which mean that there was an average of 96 accidents and 28 deaths for everyday of that year (Ogunsanya, 1991; Atubi, 2000). The situation in subsequent years has not been any better. The number of people killed in road accidents between 1990 and 2005 rose from 28,253, and the fatality rate remains consistently high (Atubi, 2009).

International comparison indicates that the chance of a vehicle killing someone in Nigeria is 47 times higher than in Britain (Atubi, 2009. The proportion of fatalities to injuries reported is also very high. For example, while Czech Republic has only one death in 175 accidents, France, one death in 175, South Africa, one death in 47 accidents, Nigeria has one death in 2.65 accidents (Atubi, 2010).

Various road safety strategies and counter measures have been used at different stages of network development. This method of seeking to prevent road accident mainly involves conscious planning, design and operations of roads. One of the most important factors in this method is the systematic identification and treatment of hazardous locations.

International comparison indicates that the chance of a vehicle killing someone in Nigeria is 47 times higher than in Britain. The proportion of fatalities to injuries reported is also very high. For example, while Czech Republic has only one death in 197 accidents, France one death in 175, South Africa, one death in 47 accidents, Nigeria has one death in 2.65 accidents (Atubi, 2010).

Road traffic accidents' statistics in Nigeria reveal a serious and growing problem with absolute fatality rate and casualty Figureure rising rapidly. In majority of developing countries, accident occurrence and related deaths are relative to either population or number of vehicles. Ironically, in Nigeria, studies have indicate that better facilities in terms of good quality and standardized roads have been accompanied by increasing number of accidents (Onakomaiya, 1988; Gbadamosi, 2002). This is totally contrary to the trends in countries were even the level of sophisticated road network and volume of vehicular traffic are much higher (Atubi, 2010).

In an effort to check this alarming trend, the Nigerian Federal Government inaugurated the Federal Road Safety Commission (FRSC) in 1988. The commission's functions include among others, the regular patrol of the highways with the aim of checking reckless driving.

3 Related Literature

Injuries are a major cause of mortality worldwide, causing more than five million deaths each year (Holder et al., 2001). As in many other areas of public health, major differences exist in countries' capacities for injury surveillance, generally corresponding to their overall economic development.

The Dutch Injury Surveillance System provides a basis for priority-setting in injury control in the Netherlands, for obtaining information on the direct medical costs of injury, and for identifying research priorities (Mulder et al., 2002).

Various states in the United States have their road traffic accident injury surveillance systems, a notable one among these is The State-wide Integrated Traffic Records System (SWITRS) used by the Department of Transport, California. SWITRS is a centralized accumulation of data for fatal and injury motor vehicle traffic accidents. In addition, a large proportion of the reported property damage only accidents are also processed into SWITRS. The reports are generated by over 500 city police departments, sheriff's offices and other local jurisdictions.

In Australia, the National Injury Surveillance Unit of the Australian Institute of Health and Welfare uses mortality statistics and hospital discharge data to produce reports on major causes of injury morbidity and mortality (Steenkamp, 2001), and the National Injury Surveillance Unit has worked in conjunction with injury surveillance and prevention practitioners in Australia to develop a set of national data standards for injury surveillance. In addition, Australia developed the National Coroners Information System, an Internet-based data storage and retrieval system for coroner cases. A state-based system, the Victorian Injury Surveillance and Applied Research System, accesses data from death certificates, coroner records, hospital admissions, and emergency department visits to conduct state-wide surveillance of injury mortality and morbidity (Watson & Ozanne-Smith, 2000).

In New Zealand, the Injury Prevention Research Unit was established in 1990 (Chalmers & Langley, 1999). The Injury Prevention Research Unit uses data files on deaths and public hospital discharges from the National Minimum Data Set compiled by the New Zealand Health Information Service to publish fact sheets and results of descriptive and evaluative studies on priority injury prevention issues (Dow et al, 2001; Langley & Smeijers, 1997; Smith & Langley, 1998). The Injury Prevention Research Unit also maintains the National Injury Query System, an Internet-accessible, menu-driven source of information on injury mortality and morbidity statistics (National Injury Query System, 2000).

In The Global Burden of Disease, Murray and Lopez noted that “still very little is reliably known about causes of death in much of the developing world” (Murray & Lopez, 1996). Although injury mortality and morbidity have been identified as major problems in some underdeveloped countries (Graitcer, 1992), injury surveillance is problematic because of the dearth of resources for public health activities in general. The World Health Organization and the CDC have recently developed a manual to help design, establish, and maintain injury surveillance systems, aimed in particular at persons working in settings with severe constraints on the capacity to keep records or assemble data into statistics (Holder et al., 2001).

In Asia, some countries have begun or are beginning to establish national systems for road injury surveillance. In other countries without such systems, surveys and studies have used existing data sets to describe and quantify data on important injury issues including road traffic injuries. Thailand initiated a provincial injury surveillance system in 1993, with reporting from five large hospitals located in Bangkok and four regions of the country. The data have been useful in documenting the large proportion of injuries from transport accidents and the large proportion of deaths occurring prior to hospitalization. The latter may indicate a need for improved prehospital transport and care. Hospital-based sentinel surveillance is being considered for a larger national injury surveillance system for the country (Santikarn, 1999).

While acknowledging this important limitation, a study in Uganda demonstrated the feasibility of a hospital-based system using a minimum data set and a simple standard index of injury severity (Kobusingye and Lett, 2000). The Caribbean Epidemiology Center of the World Health Organization is working with several member countries to develop emergency department-based injury surveillance. Recognizing the limited resources (money, personnel, etc.) available, the designers have emphasized simplicity, but they acknowledge that “even if the system is simple and flexible, sustainability will be difficult to achieve” (Ezenkwele & Holder, 2001).

Other monitoring systems include the National Injury Mortality Surveillance System in South Africa, Transport Canada’s National Collision Database (Canada), WreckWatch (White et al., 2008), OnStar (General Motors) and BMW’s Automatic Crash Notification System.

BMW’s Automatic Crash Notification System or GM’s OnStar, notify emergency responders immediately by utilizing built-in cellular radios and detect car accidents with in-vehicle sensors, such as accelerometers and airbag deployment monitors. Sensors attached to the vehicle use a built-in cellular radio to communicate with a monitoring centre that is responsible for dispatching emergency responders in the event of an emergency.

WreckWatch, a project undertaken through the collaboration of Virginia Tech and the University of Vanderbilt uses smartphones to measure the forces experienced by a vehicle and its occupants to provide a portable “black box” data recorder, accident detection system, and automatic emergency notification mechanism. (White et al., 2008).

4 Methodology

In order to develop the system, variables needed to develop a road traffic accident monitoring system were identified and reequired data were collected. The data were collected from the Osun State Sector Command of the Federal Road Safety Commission (FRSC) situated along the Gbongan-Osogbo expressway, Osogbo. The data collected includes information about the route, date of accident, description of the vehicle(s) involved, the cause of the accident, environment, collision type, nature of injury sustained by the affected road users, hospitals referred to, and health status.

In order to develop the prototype, an accident monitoring database would be developed using MySQL. In the process of developing the road traffic accident database, different tables, files, records and fields will be created. As a result, different road traffic accident factors such as the route, the date of the accident, description of the vehicle(s) involved, the cause of the accident, environment, collision type, nature of injury, hospitals referred, and health status will be stored in the database.

WampServer will be used to write MySQL queries for populating the database. The prototype will be implemented using Adobe Dreamweaver; Apache will be used as the web server to provide the basic functionality of the monitoring system. PHP will be used as a scripting language to program the server-side manipulation of the knowledge in the database. The model will be validated using data collected from the Federal Road Safety Commission (FRSC) office.

4.1 Use Case Scenario for the Road Traffic Accident Monitoring System

Use case scenarios were used to describe the interaction between users and the monitoring system. These use case scenarios were used to present the system requirements of the road traffic accident monitoring system that would make use of the developed data model. Using the scenarios afforded the opportunity to obtain the realistic description of the workflow of the system, which was to explicitly describe the intentions and actions of the users as in Jacobson (1992). The system requirements were presented with the use case scenarios to show how the road traffic accidents monitoring system would work in practice. The use case scenarios of the developed road traffic accident monitoring system are presented in Tables 1, and 2.

In the data insertion section of the developed road traffic accident monitoring system, road safety officers would be able to input the road accident information pertaining to a particular route into the system. Information such as route, collision type, crash severity, date, cause, distance from nearest hospital, and casualty/survivor count would be typed into the road traffic accident monitoring system.

In addition, the data query section of the road traffic accident monitoring system would allow users to perform various functions. One of such functions is that it allows viewing, editing, and deleting past accident records of the selected route.

Table 1: Data Insertion Use Case

Use Case Name	**Data Insertion**
Description	Scenario to illustrate data insertion for road traffic accident monitoring
Actors	User, Database Administrator
Pre-conditions	Click on hyperlink of a state followed by the desired route
Scenario	1. System checks user identification • If user is valid, system displays menu for entering accident details • If user is not valid, the system prompts for user registration 2. User enters the details of the accident including the route, collision type, crash severity, date, cause, casualty/survivor count, and distance to the nearest hospital 3. Administrator inspects, modifies, views or deletes inserted entry

Table 2: Query Use Case for Road Traffic officers

Use Case Name	**Data Query**
Description	Scenario to illustrate data query for road traffic accident monitoring
Actors	User, Database Administrator
Pre-conditions	Click on hyperlink of a state followed by the desired route
Scenario	1 System checks user identification • If user is valid, system displays menu for entering accident details • If user is not valid, the system prompts for user registration 2 User can view, modify, or delete entries captured for accidents on a selected route. 3 User can generate monthly and annual reports of reported accident occurrences

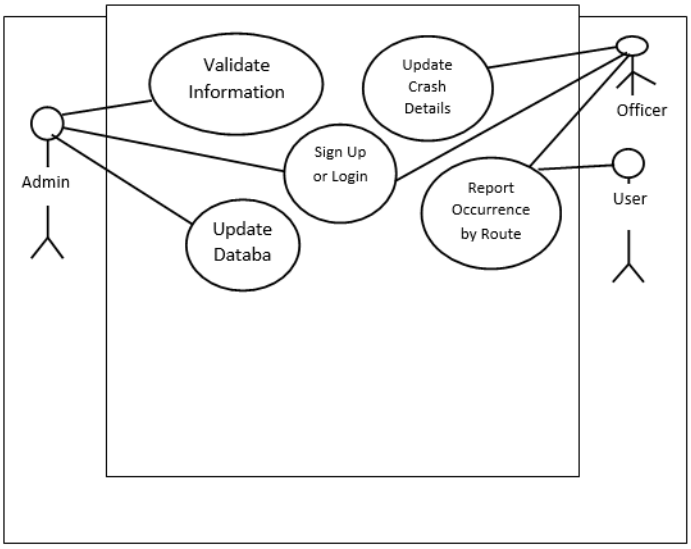

Figure 1: The Data Insertion Use Case

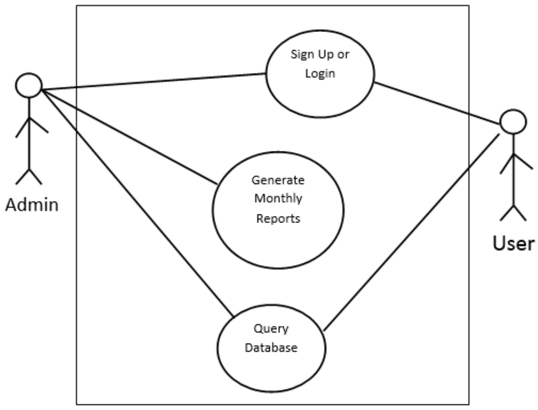

Figure 2: The Data Query Use Case

In the design of this system, there is need for a system architecture. In the developed road traffic accident monitoring system, there is need to make the design of the system easy, flexible and portable as much as possible. This was needful in order to allow users to easily make use of internet enabled mobile phones, tablets and other mobile devices, laptops, and desktop computers with web browsers to access the developed system.

The client-server system architecture was used and it is the thin client-server architecture. The road traffic monitoring system has two components namely: The server side and the client side. In the client approach almost all the work is done on demand at the server end and the client's task is to display data and information on the screen. In the thin client-server architecture, the web browser is the client.

The architecture was used because with it users will not be required to install any software on their systems. This is because a standard web browser often comes bundled with most operating systems and almost all current internet enable mobile phones and high end smartphones.

Clients would also not be required to have very powerful computer systems or mobile devices; however, the servers will require computer systems with higher conFigureuration for optimal performance. This is important as it would be regularly subjected to heavy computing tasks. There will be HTTP server and database server. Figure 1 depicts the road traffic accident monitoring system architecture.

Based on the developed system, road safety officers can make use of even mobile devices to log in to add, modify, delete or view data in the developed road traffic accident monitoring system.

4.2 The System Architecture

In the design of the system, the use of a system architecture is highly important. The design was made to be as flexible, portable and easy to use as possible. This will make it possible for users with internet-enabled mobile phones or smartphone as well as users of desktop and laptop computers to access the system.

In the Road Traffic Accident Monitoring System (RTAMS), the thin client-server architecture was used. The thin client-server architecture makes us of the Web browser as the client. This architecture makes it possible to use any internet-enabled device with a Web browser to access the system; this is in order to ensure that the system is easily available and yet cheap to implement. In this architecture, most of the processing duties is assigned to the server; the client's duty is to display the processed data and information on the screen, which in this case is the Web browser.

This RTAMS system supports a database, business logic and user interface as the major areas of design. The User Machine is the device used to access the pages and forms used for the web application, e.g. phones and personal computers etc. The Web Server is the program which allows the application to run and behave as though it is hosted on the internet, e.g. WAMP server, XAMPP, Apache etc. The Back-End is the web server and the database management system (DBMS) that holds and manages the data pool used by the application. The PHP Script controls the exchange of data between the front-end and the application back-end.

The road safety officer on the road, upon having an accident alert, immediately relays road traffic accident related information to the officer in the office so that immediate action could be taken. Also, the Road users play a very important role as they also submit accident reports by submitting reports on the RTAMS website.

Finally, based on the information available in the system, the policy makers, who usually are government officials can make rules and regulations that affect road users and all other stakeholders.

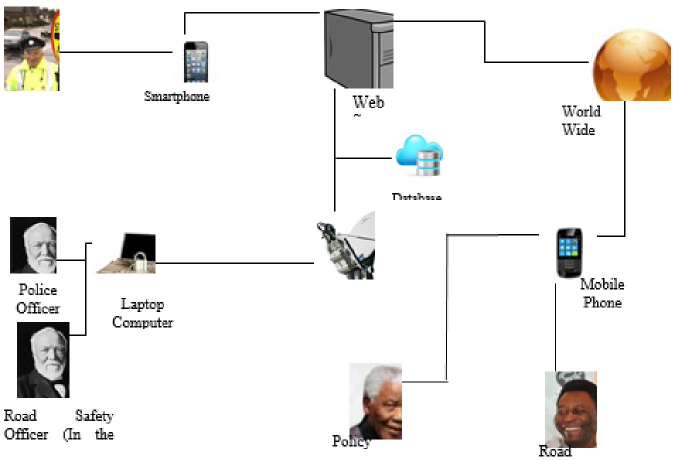

Figure 3: The System Architecture

5 Prototype Implementation

The user interface allows the user to navigate the system and also facilitate interact with the database. The system is simple and user friendly through the use of Windows Interface Menu (WIMP), and pointing devices which is very important in computer graphics design and architecture. Hence, the road tracking information system was designed to accommodate users with varying skills and competence in the area of computer usage. So with the employment of WIMP paradigm and the use of pointing devices and graphical icon that represents the specific task that the user may want to perform, makes the system easy and convenient for all. A well designed and simple user interface provides user with a better understanding of the system.

5.1 The Homepage

This part of the user interface holds all the navigations of the application. This index page contains links to help existing users of the Road Traffic Accident Monitoring System (RTAMS) to log in and new users to register. There are also news relating to road safety and accident prevention on the homepage. On the homepage, the user is able to submit accident report, new users are able to register and returning users are able to login.

5.2 The Registration Page

On the Report page, the intended user of the system is required to provide his/her username, password, Full name, phone number and email address. Upon entering these details, the data will be sent into the database. All fields are required for successful registration and if one field is left empty, a prompt will come up.

Figure 4: The RTAMS Homepage

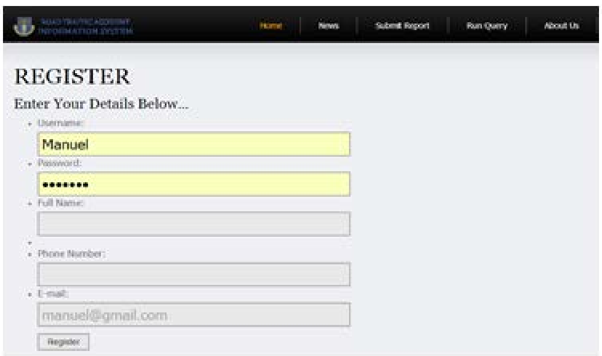

Figure 5: The Register Page

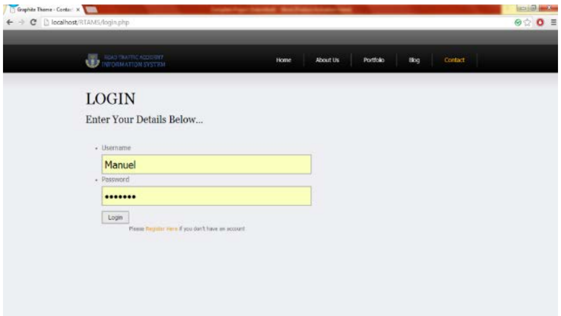

Figure 6: The RTAMS Login Page

Figure 7: Admin Dashboard

5.3 The Login Page

When the user of the system encounters the login page, he will be required to enter his username and his password to be able to log into the system. For a login attempt to be successful, the username and password combination input by the user must correspond to values that are available in the database.

5.4 The Admin Dashboard Page

On the Admin Dashboard Page, the user has the ability to submit a report, query the database, view profile, go to the homepage or logout.

5.5 "Report Accident" Page

The report page allows the user to interact with the site administrator by reporting the current state of victims and property involved in an accident as he travels along a particular route. This report is therefore sent to the incidence table of the data base, where it is checked and validated by the administrator and then updated into the various fields of the data base for appropriate report generation.

The road's users are allowed to also report the accident occurrences on a particular road at any point in time. The road user can also make reports even if he does not have an account. The user clicks on the "Chat with Officer" link on the homepage and starts a chat with an office; he will be required to enter his name, his phone number, e-mail address and the description of the accident. This information is sent to the database where it will be validated by the administrator and finally updated. Below are screenshots of the Road Traffic Accident Monitoring System (RTAMS).

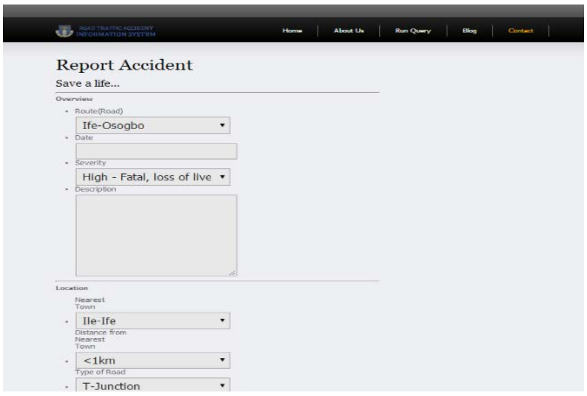

Figure 8: The Report Submission Page

Figure 9: The Database Query Page

5.6 The Database Query Page

The user of the system has the ability to query the system based on three variables – The route (Ife-Osogbo, Osogbo-Ilesa or Ilesa-Ife route), severity (High, medium, or low) and Names (which would be entered in the textfield).

5.7 The Individual Report Page

This page gives information about individual accident reports based on location (Nearest town, distance from nearest town, type of road, road surface condition, weather condition, light condition), vehicle information (vehicle type, number plate, vehicle condition, and vehicle colour), Accident Details (Date of Occurrence, Vehicle count, Total Victim count, uninjured victim count, injured victim count and dead victim count). Also, details about identification documents that were found at the accident scene are also entered on this page and finally, details that pertain to first aid (nearest town, nearest hospitals, and types of injuries) will also be entered on this page.

5.8 Contact Page

On the contact page, any user with complaints or information can use this page to send message to the system administrator or to road safety officers. The sender is required to enter his name, email address, the subject of the message and the message body.

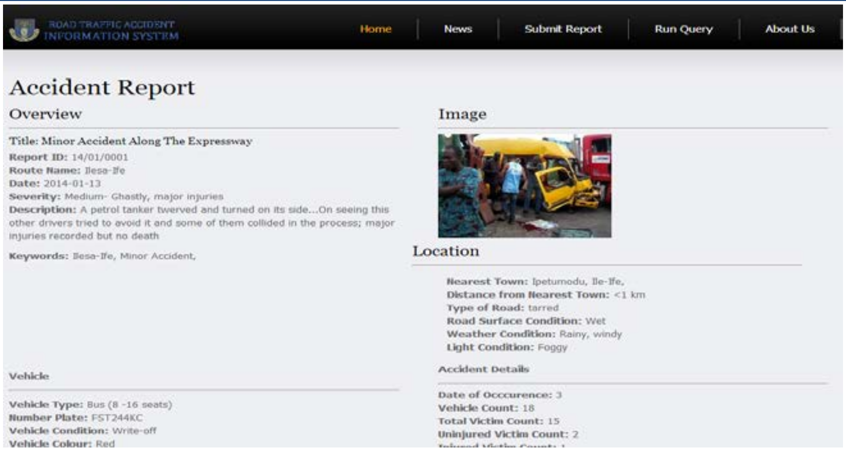

Figure 10: The Accident Report Page

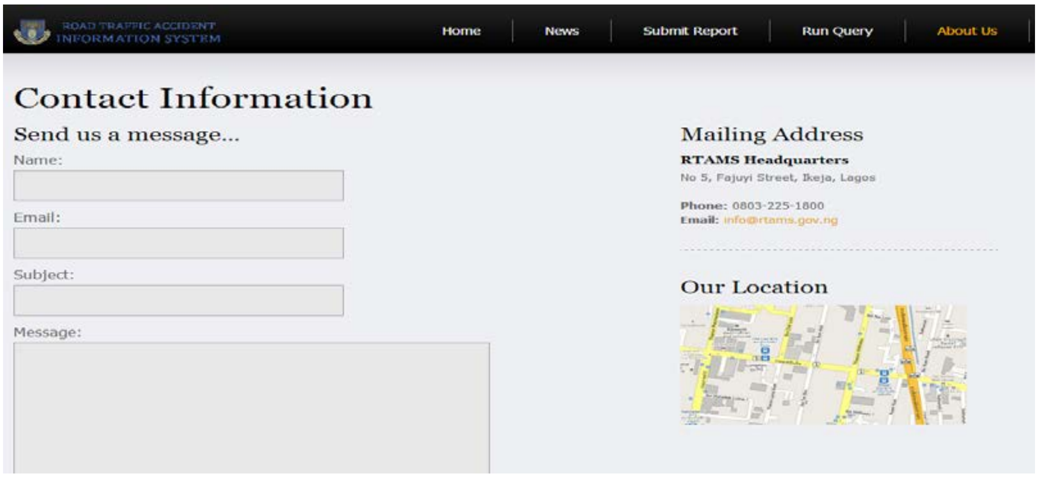

Figure 11: Contact Page

6 Conclusion

This paper focuses on the monitoring of road traffic accident occurrences in order to derive useful data such as the severity of the accident, the types of vehicles and victims involved in the accidents, the type of injuries sustained (in order to facilitate adequate first aid preparations), the condition of the road, the weather condition, the light condition and any form of identification detail gotten from the accident scene.

This paper shows an effective way of monitoring road traffic accident information which is a semi-automatic monitoring system in that it requires human interactions in the area of data and observation collection and updating of the database.

REFERENCES

[1] Akinyemi, E. O. (2009). Contributing Road Factors in Accidents on Rural Roads in Nigeria, in Road Traffic Accidents in Developing Countries Vol. I Asalor, Onibere and Ovuworie – eds, Joja press, Lagos.

[2] Alban (2011) War on the Roads. Basic Medical Journal. Vol. 324, No. 7346.

[3] Allaramu, (2011) “Further Research on Accident Rates in Developing Countries”. Accident Analysis and Prevention. Vol. 18, No. 2, pp. 119-127.

[4] Amber (2003). Accident Mortality and Morbidity in Developing Countries. In Manciaux M. and Romer, C.J. (ed) Accidents in Childhood and Adolescence: The Role of Research, Pp. 31-38. Geneva: World Health Organisation.

[5] Annest, J.L., Pogostin, C.L. (2000) CDC’s short version of the ICECI. International classification of external causes of injury: a pilot test. Report to the World Health Organization collaborating centers on the classification of disease. Atlanta, GA: Office of Statistics and Programming, National Center for Injury Prevention and Control, Centers for Disease Control and Prevention, 2000. (http://www.cdc.gov/ncipc/pub-res/icecibk.pdf).

[6] Asalor, J. O. (2010). Towards Improved Road Safety in Nigeria. Technical Report No. Rts/00/82/011, Faculty of Engineering, Uni of Benin.

[7] Atubi, A.O. (2010) Spatial and Temporal Perspective on Road Traffic Accident variations in Lagos mainland, South Western Nigeria. African Research Review,Vol.4 (1) pp.256-272.

[8] Atubi, A.O. (2010) Road Transport System Management and Traffic in Lagos, South Western Nigeria, Journal of African Research Review.Vol. 4. No.4. Pp. 459-470.

[9] Atubi, A.O. (2012) Determinants of road traffic accidentoccurrences in Lagos State: Some Lessons for Nigeria. International Journal of Humanities and Social Science. Vol. 2, No. 5, Pp. 225-343.

[10] Atubi, A.O. (2012) Epidemiology of Injuries from Road Traffic Accidents in Lagos State, Nigeria. International Journalof Science and Technology, Vol. 1, No. 2, Pp. 56-75.

[11] Atubi, A.O. and Onokala, P.C. (2009) Contemporary Analysis of Variability in Road Traffic Accidents in Lagos State, Nigeria. Journal of African Geographical Review. Vol. 28, Pp. 11-41.

[12] Atubi, A.O.(2009) Modelling Road Traffic Accidents in Lagos State, South Western Nigeria. Journal of Society and StateVol. 1. Nos 1 &2, pp 57-74.

[13] Aworemi, A. and Alawode, E.C. (2009) Roads to where? Newswatch, December 30, 17 (13) 3.

[14] Balogun (2001) “Innovative Options for Financing Transport Infrastructure in Nigeria” In Nisereel, The Magazine of the Nigerian Institute of Social and Economic Research, NDS485, Ibadan

[15] Blandford, A., and William Wong, B.L. (2004) Situation awareness in emergency medical dispatch. International Journal of Human-Computer Studies, 61(4):421–452,.

[16] Blond, L., Hansen, L.B. (1999) Injuries caused by falling soccer goalposts in Denmark. Br J Sports Med 1999;33:110–13

[17] Central Bank of Nigeria (1994) Annual Report and Statement of Account for the Year Ended 31 December, 1993. C.B.N., Lagos April.

[18] Central Bank of Nigeria (1997) Annual Report and Statement of Account for the Year Ended 31 December, 1996. C.B.N., Lagos April.

[19] Centre for Black Culture and International Understanding (2011) Osun State in Brief http://www.centreforblackculture.org/osunstate.htm. Retrieved 17th June 2013.

[20] Chalmers, D.J, Langley, J.D. (1999) New Zealand's injury prevention research unit: helping shape injury prevention policy and practice. Inj Prev 1999;5:72–5.

[21] Committee on Environmental Epidemiology National Research Council. (1997) Injury in America—a continuing public health problem. Washington, DC: National Academy Press.

[22] Conallen (2004). Driver Training of Africa: the UN-ECA Driving Manual. Proceedings of the Second African Road Safety Congress. Oslo: Institute of Transport Economics (TOI).

[23] Department of Health and Children. EHLASS report for Ireland 2000. Dublin, Ireland: European Home and Leisure Accident Surveillance System. (http://europa.eu.int/comm/health/ph/programmes/injury/outcome2000/fp_injury_2000_exs_19_en.pdf).

[24] Dimitrijevic et al., (2007). Road accidents and traffic flows: An econometric investigation. In Economica 67, pp 101-21.

[25] Dow NA, Stephenson SCR, Allnatt DM. Trends in thermal injury. Burns from fire or flames. Dunedin, New Zealand: Injury Prevention Research Unit, University of Otago, 2001. (Fact sheet no. 21) (http://www.otago.ac.nz/IPRU/FactSheets/FactSheet21.pdf).

[26] Dow, N.A, Stephenson, S.C., Casey, D.M. (2001) Injury to children in New Zealand resulting in death or hospitalization. Dunedin, New Zealand: Injury Prevention Research Unit, University of Otago, (Fact sheet no. 22) (http://www.otago.ac.nz/IPRU/FactSheets/FactSheet22.pdf).

[27] Endsley, M.R. (1995) Toward a theory of situation awareness in dynamic systems. Human Factors: The Journal of the Human Factors and Ergonomics Society, 37(1):32–64.

[28] E-Nigeria (2011) About Osun State http://www.e-nigeria.net/osun.html. Retrieved 17th June 2013

[29] European Union Road Federation, ERF (2001) Comparative Accident Costs in Developing Countries. TRRL Supplementary Report 206. Crowthorne: Transport and Road Research Laboratory.

[30] Expert Vanguard (2011). Work and health country profiles: Country profiles and national surveillance indicators in occupational health and safety (People and Work Research Reports 44).

[31] Ezenkwele, U.A, Holder, Y. (2001). Applicability of CDC guidelines toward the development of an injury surveillance system in the Caribbean. Inj Prev; 7:245–8.

[32] Ezenwa, A.O. (1986) "Prevention and Control of Road Traffic in Nigeria" Journal of the Royal Society of Health. Vol. 106, No. 1, Pp. 25-26.

[33] Federal Road Maintenance Agency (2001) An Empirical Investigation into Nigeria Road Accident Causation Factors Annual Report and Statement of Account

[34] Filani, M.O. and Gbadamosi, K.T. (2007) Spatial and Temporal Pattern of Road Traffic Accident Occurrences in Nigeria: 1970-1995. Nigerian Geographical Journal, Vol. 5, No. 1, Pp. 55-70.

[35] Fingerhut, L.A, Cox, C.S, Warner, M. (1998) International comparative analysis of injury mortality. Findings from the ICE on injury statistics. International Collaborative Effort on Injury Statistics. Adv Data; (303):1–20.

[36] Foster, Warren G., Agzarian, John (2006). "Reporting results of biomonitoring studies". Analytical and Bioanalytical Chemistry 387 (1): 137–40. doi:10.1007/s00216-006-0822-6. PMID 17093961.

[37] Fouracre, P. R. and Jacobs, G.D. (1976) Comparative Accident Costs in Developing Countries. TRRL Supplementary Report 206. Crowthorne: Transport and Road Research Laboratory.

[38] Gbadamosi, K.T. (1994) "Spatial Analysis of Road Accidents in Nigeria, 1970-1990" Unpublished M.Sc. Thesis, Ogun State University, Ago-Iwoye, Nigeria.

[39] Graitcer, P.L. (1992) Injury surveillance in developing countries. MMWR CDC Surveill Summ; 41:15–20.

[40] Gungul, T. (2012). Road Traffic Accidents in Nigeria: Causes and Control, Journal of Applied Science and the Environment, Vol 3, pp 25-29.

[41] Haddon, W. (1980) Advances in the Epidemiology of Injuries as a basis for public policy. Public Health Reports. 95: pp. 411-421.

[42] Harrald, J. and Jefferson, T. (2007) Shared situational awareness in emergency management mitigation and response. In System Sciences, 2007. HICSS 2007. 40th Annual Hawaii International Conference on, page 23. IEEE.

[43] Hoberman (2009). An Exploratory Simulator Study on the Use of Active Control Devices in Car Driving. TNO-Report IZF 1992 B-2., TNO Institute for Perception. Soesterberg, The Netherlands.

[44] Holder, Y., Peden, M., Krug, E, et al, eds. (2001) Injury surveillance guidelines. Geneva, Switzerland: World Health Organization,. (Document no. WHO/NMH/VIP/01.02) (http://www5.who.int/violence_injury_prevention/download.cfm?id=0000000136).

[45] Injury Section, Population and Public Health Branch, Health Canada. 2003 Injury surveillance on-line. Tunney's Pasture, Ottawa, Canada: Canadian Hospitals Injury Reporting and Prevention Program, Health Canada. (http://www.hc-sc.gc.ca/pphb-dgspsp/publicat/chirpp-schirpt/index.html).

[46] International Collaborative Effort (ICE) on Injury Statistics. Injury ICE activities. Hyattsville, MD: National Center for Health Statistics, Centers for Disease Control and Prevention, 2002. (http://www.cdc.gov/nchs/about/otheract/ice/projects.htm).

[47] Jacobs, G.D.; Aaron – Thomas, A; Astrop, A. (2000) Estimating Global Road Facilities. Crow Thorne, U.K. Transport Research Laboratory Report 445. Pp. 1-35.

[48] Jacobs, G.D. and Sayer, I.A. (1983) Road Accidents in Developing Countries'. Transport and Road Research laboratory, TRRI Supplementary Report 897 Crowthorne.

[49] Jacobson (1992). Estimating Global Road Facilities. Crow Thorne, U.K. Transport Research Laboratory Report 445. Pp. 1-35.

[50] Keuffel (1996). Finnish occupational exposure databases. Applied Occupational and Environmental Hygiene, 16(2), 154–158.

[51] Kobusingye, O.C, Lett, R.R. (2000) Hospital-based trauma registries in Uganda. J Trauma; 48:498–502.

[52] Krug, E.G., Sharma, G.K.; Lozano, R. (2000). The Global Burden of Injuries. American Journal of Public Health. 90(4); 523-526.

[53] Langley, J.D., Smeijers, J. (1997) Injury mortality among children and teenagers in New Zealand compared with the United States of America. Inj Prev; 3:195–9.

[54] Langley, J.D. (1995) Experiences using New Zealand's hospital based surveillance system for injury prevention research. Methods Inf Med; 34:340–4.

[55] Le, L., Pham, C., Linnan M., et al. (2002) Vietnam profile on traffic-related injury: facts and Figure from recent studies and their implications for road traffic injury policy. Presented at the Road Traffic Injuries and Health Equity Conference, Cambridge, Massachusetts, April 10–12,

[56] Li G.H., Baker S.P. (1991) A comparison of injury death rates in China and the United States, 1986. Am J Public Health 81:605–9.

[57] Lu and Lu (2004). Motor Vehicle Injuries in Taiwan. Asia Pacific Journal of Public Health, 4:72-75.

[58] Lyons, R. A., Brophy, S., Pockett, R., et al. (2005). Purpose, development and use of injury indicators. Int J Inj Contr Saf Promot 12(4): 207-11

[59] Mackenzie SG, Pless IB. (1999) CHIRPP: Canada's principal injury surveillance program. Inj Prev; 5:208–13.

[60] Mattord, Verma (2008). Principles of Information Security. Course Technology. pp. 290–301. ISBN 978-1-4239-0177-8.

[61] Medical Research Council and UNISA. (2007). A profile of fatal injuries in South Africa 7th Annual Report of the National Injury Mortality Surveillance System 2005.MRC/UNISA Crime, Violence and Injury Lead Programme, July 2007.

[62] Mock CN, nii-Amon-Kotei D, Maier RV. (1997) Low utilization of formal medical services by injured persons in a developing nation: health service data underestimate the importance of trauma. J Trauma; 42:504–11.

[63] Mohan, P., Padmanabhan, V.N., and Ramjee, R.. (2008) Nericell: Rich monitoring of road and traffic conditions using mobile smartphones. In Proceedings of the 6th ACM conference on Embedded network sensor systems, pages 323–336. ACM New York, NY, USA.

[64] Mulder S. (2001). Surveillance and priority-setting: where to start in preventing home and leisure accidents? Ridderkerk, the Netherlands: Ridderprint.

[65] Mulder, S., Blankendaal, F., Vriend, I., et al. (2002). Epidemiological data and ranking home and leisure accidents for priority-setting. Accid Anal Prev; 34:695–702.

[66] Murray, C.J., Lopez A.D, eds. (1996) The global burden of disease: a comprehensive assessment of mortality and disability from diseases, injuries, and risk factors in 1990 and projected to 2020. Geneva, Switzerland: World Health Organization.

[67] Mutabazi, (2011) Equity Dimensions of Road Traffic Injuries in low and Middle-income countries. Injury Control and Safety prevention. 10(1-2): 13-20.

[68] Najm, W. G., Sen, B., Smith, J. D., & Campbell, B. N. (2003). Analysis of Light Vehicle Crashes and Pre-crash Scenarios based on the 2000 General Estimates System. Washington D.C.: U.S. Department of Transportation, report no. DOT-VNTSC-NHTSA-02-04.

[69] The National Coroners Information System—a death and injury prevention tool. J Law Med 2000; 8:155–6.

[70] National Injury Mortality Surveillance System. (2009). Website: (http://www.sahealthinfo.org/violence/nimss.htm , accessed 08 October, 2009)

[71] National Injury Query System. Dunedin, New Zealand: Injury Prevention Research Unit, University of Otago, 2000. (http://www.otago.ac.nz/ipru/Statistics/NIQS.html).

[72] National Road Safety Council of Kenya (1992) Accident Statistics, 1983-1990. Nairobi: Ministry of Public Works, Government of Kenya. Onokala, P.C. (1995) The Effect of Land Use on Road Traffic Accident in Benin-City, Nigeria. Journal of Transport Studies, Vol. 1, No. 1, Pp. 34-44.

[73] Norman, R, Matzopoulos, R, Groenwald, P and Bradshaw, D. (2007). "The high burden of injuries in South Africa." Bulletin of the World Health Organization.September 2007, 85 (9). WHO. Geneva.

[74] Nwaneri (2011) General Trend of Safety and Accident Records in Nigerian Transport Sector. In: Bolade T. and Ogunsanya, A. (eds) Accident Control and Safety Measures in Mass Transit Operation in Nigeria, Ibadan University Press, P. 11.

[75] Odugbemi, O. O. (2010). Road Transportation and Tourism in Nigeria, Joja Press, Lagos.

[76] Opawole et al., (2011). General Trend of Safety and Accident Records in Nigerian Transport Sector. In: Bolade T. and Ogunsanya, A. (eds) Accident Control and Safety Measures in Mass Transit Operation in Nigeria, Ibadan University Press, P. 11.

[77] Organization for Economic Cooperation and Development Organization (2004) Managing the Environment: the Role of Economic Instrument. Paris

[78] Oluwasanmi, A.J. (1993) Road Accident Trends in Nigeria. Accident Analysis Prevention 25 (4) Pp. 485-487.

[79] Onokala, P.C. (1995) The Effect of Landuse on Road Traffic Accident in Benin-City, Nigeria. Journal of Transport Studies, Vol. 1, No. 1, Pp. 34-44.

[80] Oyeyemi (2003) Road Accident Trends in Nigeria. Accident Analysis Prevention 25 (4) Pp. 485-487.

[81] Ozanne-Smith, J., Ashby, K., Stathakis, V., et al. (2005) A decade of Victorian injury surveillance. In: Hazard. 40th ed. Victoria, Australia: Victorian Injury Surveillance and Applied Research Function, Monash University Accident Research Centre, 1999:1–9. (http://www.general.monash.edu.au/muarc/hazard/haz40.pdf).

[82] Peden, M., Toroyan, T. Counting road traffic deaths and injuries: poor data should not detract from doing something! Ann Emerg Med; 46: 158-60

[83] Phillips, M.R, Li, X., Zhang, Y. (2002) Suicide rates in China, 1995–99. Lancet; 359:835–40.

[84] Quigley M, Chandramohan D, Setel P, et al. (2000) Validity of data-derived algorithms for ascertaining causes of adult death in two African sites using verbal autopsy. Trop Med Int Health; 5:33–9.

[85] Rauscher S., Messner G., Baur P., Augenstein J., Digges K., Perdeck E., Bahouth G., and Pieske O. (2009) Enhanced Automatic Collision Notification System- Improved Rescue Care Due To Injury Prediction- First Field Experience.

[86] Ross, A. Baguley, C. Hills, V. Mchonald, M. and Silcock, d. (1991) Towards Safer Roads in Developing Countries: A guide for Planners and Engineers. Crow Thorne, U.K.; Transport Research laboratories.

[87] Santikarn C, Punyaratabandhu P, Podhipak A, et al. (1999) The establishment of injury surveillance in Thailand. Int J Consumer Product Saf; 6:133–43.

[88] Simsion and Witt (2005). Automotive Collision Avoidance System Field Operational Test: Warning Cue Implementation Summary Report. National Highway Transportation Safety Administration Report No. DOT HS 809 462.

[89] Scarfone, Karen; Mell, Peter (2007). "Guide to Intrusion Detection and Prevention Systems (IDPS)". Computer Security Resource Center (National Institute of Standards and Technology) (800–94).

[90] Short, Nicholas, M. (2009) "Remote Sensing Tutorial." U.S. National Aeronautics and Space Administration (NASA). Greenbelt, MD.

[91] Smith, G.S, Langley J.D. (1998) Drowning surveillance: how well do E-codes identify submersion fatalities? Inj Prev; 4:135–9.

[92] Stansfield, S.K.; Smith, J.S. and McGreevey, W.P. (1992) Injury. In Diseases Priorities in Developing Countries. (eds) Jamison, D.T. and Mosley, W.H. World Bank, Oxford University Press, London, Pp. 609 -633.

[93] Stauffer, C. and Grimson, W. E., (1999). "Adaptive background mixture models for real-time tracking," in Proc. CVPR, June 1999, pp. 246–252.

[94] Steenkamp, M., Cripps, R. (2001) Child injuries due to falls. Canberra, Australia: Australian Institute of Health and Welfare,. (Injury research and statistics series, no. 7) (AIHW catalogue no. INJCAT 37) (http://www.nisu.flinders.edu.au/pubs/reports/2001/falls_injcat37.pdf).

[95] Steenkamp, M., Harrison, J.E. (2001) Suicide and hospitalised self-harm in Australia. Canberra, Australia: Australian Institute of Health and Welfare. (Injury research and statistics series, no. 1) (AIHW catalogue no. INJCAT 30).

[96] Taket, A. (1986) Accident Mortality in Children, Adolescence and Young Adults. World Health Statistics Quarterly, 39:232-256.

[97] Transport and Research Laboratory (1999) Towards Safer Roads in Developing Countries, TRL, Crowthorne, U.K.

[98] Verma, M., Lange, R., and McGarry, D. (2007) A Study Of US Crash Statistics From Automated Crash Notification Data.

[99] Watson, W.L, Ozanne-Smith, J. (2000) Injury surveillance in Victoria, Australia: developing comprehensive injury incidence estimates. Accid Anal Prev;32:277–86.

[100] West and Fowler (2003).) Road accidents and traffic flows: An econometric investigation. In Economica 67, pp 101-21.

[101] White Jules, Chris Thompson, Hamilton Turner, Brian Dougherty, and Douglas C. Schmidt (2008), WreckWatch: Automatic Traffic Accident Detection and Notification with Smartphones. Journal of Mobile Networks and Applications.

[102] Witten, I. H. and Frank E. (2000). Data Mining: practical machine learning tools and techniques with java implementations. San-Francisco, Morgan Kaufmann publishers.

[103] World Bank (1990) “Road Safety Problem in the Third World”. The Urban Edge. Vol. 14, No. 1.

[104] World Bank (2004): Development Practice: Sustainable Transport Priorities for Policy Reform Chapter 3, Washington D. C.

[105] World Health Organisation (1984) Road Traffic Accidents in Developing Countries, Geneva: WHO, TRS 703

[106] World Health Organisation (1989) Analysis of Achievements of Traffic Safety in Industrialised and Developed Countries.

[107] WHO (2009). Global Status Report on Road Safety: Time for Action. Geneva. World Health Organisation www.who.int/violence_injury_prevention/road_safety_status/2009

[108] World Health Organisation (1984) Road Traffic Accidents in Developing Countries, Geneva: WHO, TRS 703

[109] Wrona, F. J.; Cash, K. J., (1996), "The ecosystem approach to environmental assessment: moving from theory to practice." Journal of Aquatic Ecosystem Health. Kluwer Academic Publishers, ISSN 0925-1014

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Predictive Model for Likelihood of Survival of Sickle-Cell Anaemia (SCA) among Peadiatric Patients using Fuzzy Logic

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ABSTRACT

A fuzzy logic-based system has been applied to a number of cases in medicine especially in the area of the development of diagnostic systems and has been discovered to produce accurate results. In this paper, a fuzzy logic-based system is presented which is used to simulate a prediction model for determining the likelihood of Sickle Cell Anemia (SCA) in individuals given a 3-tuple record containing the level of fetal haemoglobin, genotype and the degree of Anemia.

Knowledge was elicited from an expert at Federal Medical Centre, Owo, Ondo State, Nigeria and was used in developing the rule-base and simulated the prediction model using the MATLAB software. The results of the fuzzification and defuzzification of variables, inference engine definition and model testing was also presented and showed that the fuzzy logic based model will be very useful in the prediction of the likelihood of Sickle Cell Anemia (SCA) among Nigerian patients.

Keywords: fuzzy logic, prediction model, sickle-cell disease, likelihood

1 Introduction

According to Obeagu et al (2014) sickle cell disease (SCD) is a hereditary blood disorder which can be easily identified as an abnormal, sickle-shaped form of the red blood cells. A complication called sickling arises due to the cells' flexibility caused by the sickle shape and reduces the lifespan of an average male and female to 42 and 48 years respectively. If this sickling condition is well-managed, there are cases where such a person may live up to 8 decades. The disease is also discovered to be very rampant among tropical and sub-tropical sub-Saharan regions where there used to be malaria (Wellems et al, 2003). SCA may lead to various acute and chronic complications, several of which have a high mortality rate (Malowany et al, 2012). People with sickle cell disease may also develop anemia including some jaundice and body pains.

Three quarters of sickle-cell cases occur in Africa. A recent WHO report estimated that around 2% of newborns in Nigeria were affected by sickle cell anaemia, giving a total of 150,000 affected children born every year in Nigeria alone. The carrier frequency ranges between 10% and 40% across equatorial Africa, decreasing to 12% on the North-African coast and <1% in South Africa (Obeagu et al, 2014). Sickle Cell is a major cause of morbidity and mortality in Africa where there is no readily effective treatment (Omoti, 2005).Patients with sickle cell disease have varying amounts of

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abnormal haemoglobin called the sickle cell in their erythrocytes. Sickle cell anaemia is due to the substitution of adenine with thymine in the glutamic DNA codon, which results in turn, in substitution of valine for glutamic acid in position of beta globin chain (Pauling et al., 1949).

The disease amounts for over 60% of the world's major haemoglobinopathies with an estimated 2-3 million Nigerians affected by the S gene (Olatunji, 2002).The extent of the problems of sickle cell disease in Nigeria cannot therefore be overemphasized because of the S gene said to be between 25-30%. The majority of patients born to rural dwellers do not usually survive childhood (Ukpong, 1992).

Fuzzy logic is a means of providing a path for the diagnosis and decision making process due to its ability to deal with uncertainties (fuzziness) and ambiguity which may exist in the knowledge and information relating to a domain of study. Today, medical practitioners have identified possible and promising areas for implementing fuzzy logic systems for medical diagnosis Mishra et al, 2014). The idea of Fuzzy logic was presented by Lofti A Zadehn in 1965 based on the fuzzy set theory. Fuzzy logic systems are implemented by the manipulation of membership functions which simulate variables by the inference engine (rule-base).

Membership functions (MF) are curves that defines how each point in the input and output space is mapped to a membership value (or degree of membership) between 0 and 1.This implies that for every label of each variable; a membership function will be used to define the level of membership of the value entered with respect to the degree of membership to the label. Unlike, classical set; a fuzzy logic may be defined as follows:

If X is a universe of discourse and its elements are denoted by x, then a fuzzy set A in X is defined as a set of ordered pair:

$$A = \{x, \mu_A(x) |\ x \in X\} \tag{1}$$

$\mu_A(x)$ is called a membership function (or MF) of x in A. The membership function maps each element of X to a membership function value between 0 and 1. For the purpose of this study, the following must be noted:

- The set A is any input (sickle-cell factors) or output (sickle-cell likelihood) variable considered for this study;
- The set X is the set of values for which a variable is valid, for example a set A = degree of Anemia will be valid for value x=0 for No and x=1 for Yes. Hence for the set A, the set X is the set containing {0, 1}; and
- $\mu_A(x)$ is the map of the membership function that will be used to plot the degree of membership.

Furthermore, there is no widely acceptable and readily available cure for patients with sickle cell anaemia at present. Curable methods such as gene therapy and bone marrow transplantation, which may be associated with several complications, are not readily available in developing nations (Omoti, 2005). This disease is a serious threat to human life and it is believed that such tragedy can be reduced by early diagnosis of its existence, hence this study.

This paper is aimed at developing a fuzzy logic based system that predicts the likelihood of sickle cell disease in an individual by requesting for a 3-tuple record consisting of the Level of fetal haemoglobin, Genotype and the degree of Anemia. The study is limited to knowledge elicited from a physician located in western Nigeria based on experience gathered in the diagnosis of the likelihood of sickle cell disease in patients in western Nigeria.

2 Related Works

Goyal, D. (2006) worked on the development of a disease diagnostic system using LabVIEW. The work was based on the use of fuzzy logic to diagnose the various kinds of anemia using the concept of fuzzy logic in the LabVIEW platform. The proposed system made use of 12 input variables which all had their respective labels: haemoglobin units, hematocrit units HMCT), mean corpuscular volume (MCV), mean corpuscular hemoglobin concentration (MCHC), reticulocyte count (RCC), while blood cells (WBC), platelets (PLT), total Iron Binding Capacity (TIBC), serum Iron (SEI), Nucleated red cells (NRC), hyper segmented white cells (HSWC) and ringed sideroblast in bone marrow (RSBM). The system was simulated on the LABVIEW platform and a fuzzy logic system containing 12 input variables was used to predict 18 different types of anemia. After series of test, the system was discovered to produce excellent results in the diagnoses of cases collected from the laboratory.

Sayyahi (2008) worked on an application of fuzzy cased based reasoning for poison classification. The system was developed with the aim of recognizing, controlling and treating a limited poison case. The system developed for the classification of poison involved the combination of fuzzy logic and case-based reasoning – the case based reasoning system was developed which incorporated the use of fuzzy logic. The variables used as inputs were symptoms categorized as: general, psychological, cardiovascular, respiratory and gastrointestinal. The system was discovered to identify the different types of poison for which information was provided by the physician although, its results are still subject to the physician's decision to accept or reject (if he feels otherwise).

Aramideh et al (2014) worked on the application of fuzzy logic for the development of an expert system to diagnose anemia. The fuzzy model was developed using 5 input variables namely: tachycardia, irritability, memory weakness, nose bleeding and chronic fatigue. The fuzzy system also has 3 different outputs which identify 3 different types of anemia namely: iron deficiency, folic acid deficiency and sickle-cell. The inference engine of the fuzzy model contained 42 rules which were modeled using the 5 input and 3 output variables. The authors also suggest that additional symptoms of anemia could help produce a more effective model.

Mishra et al (2014) developed a fuzzy logic model using the Mamdani model for the effective diagnosis of sickle-cell disease. The model developed used rules that were generated based on support sets from patients who belong to three classes: patients with primary SCD, secondary SCA and without SCD. The diagnostic system used three input variables namely: symptom score expressed as a percentage of severity of symptoms, haemoglobin A and haemoglobin S. The results showed that the fuzzy model improves results compared to other existing models. It was also discovered to be capable, efficient and cost effective in diagnosing SCD.

3 Materials and Methods

3.1 Research design

In this paper, a fuzzy logic-based prediction model is proposed with the aim of predicting the likelihood of sickle-cell disease in individuals. In order to achieve this, the research design presented in Figure 1 was used. The study started with the identification of the problem of predicting sickle cell likelihood given a number of symptoms/factors considered as input variables (3 in all). A review of related literature was performed to identify understand sickle cell diseases and its symptoms in addition to related works done in the past. Following this, knowledge was elicited from an expert (medical practitioner) located at the Federal Medical Centre, Owo, Ondo State in understanding and verifying the information concerning sickle cell symptoms.

The elicited knowledge was used to build the inference engine of the proposed system – this is part of the model formulation technique which also includes the fuzzification of the input and output variables. the model formulation is made complete by the identification of the aggregation method chosen for the inference engine alongside the defuzzification method required for producing the output variable which is the likelihood of sickle-cell disease (No and Yes)

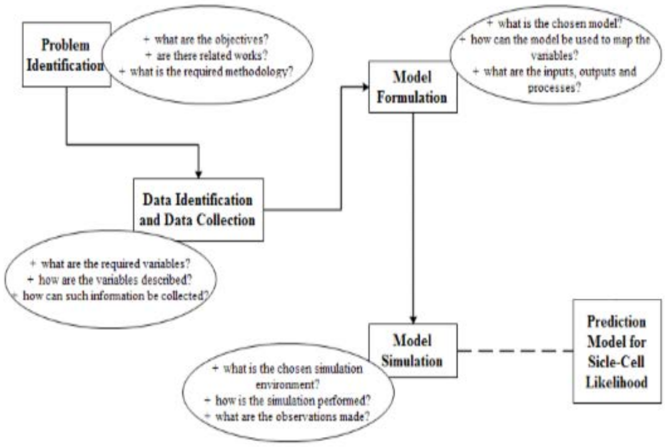

Figure 1: Research design for the study

3.2 Data identification and collection

A number of symptoms/factors are known to be connected to sickle cell disease, among all these factors only 3 were identified as being the most important and relevant symptoms: the level of fetal haemoglobin, genotype and the degree of anemia. This information was collected via structured interview with the medical practitioner who identified the factors and emphasized 3 main factors which are most easily used in identifying the likelihood of sickle cell disease based on his experience in medical practice. The fetal haemoglobin is defined as either: less than 2%, between 2% and 5%, and greater than 5%; the genotype was classified as either SS, S* and SA while the degree of anemia is classified as either less than 15% and greater than or equal to 15%.

In addition to the identification of the data variables, an understanding of the pattern of distribution was important in identifying the best membership function that could be used in plotting the labels of each variables. The number of rules required by the fuzzy logic inference system was calculated by multiplying the labels of each variable with each other; therefore we have 3*3*2 = 18 different rules. This information was necessary in the development of the fuzzy logic inference system.

3.3 Fuzzy logic model formulation

Fuzzy logic systems have the ability to decide and control a system using the knowledge of an expert. Fuzzy logic systems are mostly profitable in systems with sophisticated environments where a clear and obvious model of the system is not achievable.

In order to develop the fuzzy logic system required for the prediction of the likelihood of sickle-cell disease, a number of activities are needed to be accomplished. The Fuzzy Logic System available in the Fuzzy Logic Toolbox of the MATLAB R2012a software has three parts (see Figure 2):

- A set of Inputs represented by their respective membership functions;
- An Inference Engine which contains the IF-THEN rules (domain knowledge); and
- An Output represented by its membership functions.

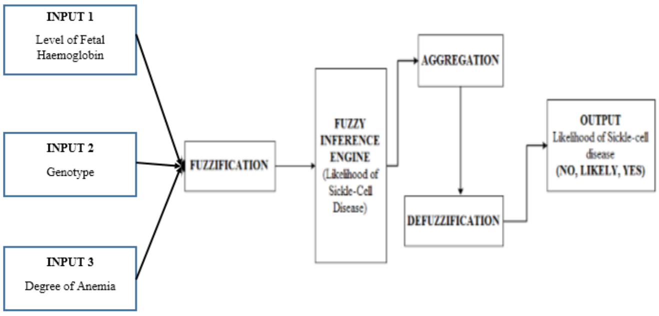

Figure 2: Schematic diagram of the proposed model

The membership functions will be used to map the values of each input and output variables into a [0, 1] interval with the use of triangular and trapezoidal membership functions (where appropriate); this process is referred to as a Fuzzification process. After Fuzzification; the fuzzified inputs must be mapped to the fuzzified output via the use of operators (AND, OR and NOT) to develop IF-THEN rules that describe the relationship between every input (sickle-cell likelihood factors) and output (likelihood of the disease) variable. The different rules are used to generate different results which are then aggregated to just one fuzzified output. This fuzzified output will then be defuzzified using the centroid method which selects the centre of the polygon to determine the label of the output variable as Yes, Likely or No.

The most prominent reasons that justify the use of fuzzy logic systems today are (Aramideh et al, 2014):

a) The sophistication of the natural world which leads to an approximate description or a fuzzy system for modeling; and
b) The necessity of providing a pattern to formulate mankind knowledge and applying it to actual systems.

The process of development of the fuzzy inference system needed for the prediction of sickle-cell disease may summarized as follows:

i. Fuzzification of inputs and outputs;
ii. Construction of the inference engine;
iii. Rule aggregation; and
iv. Defuzzification of output variables.

3.4 Defining membership functions

Before the process of Fuzzification, it is very important to properly describe the crisp values that was used in mapping the values of the membership function which was be needed by the fuzzy logic system. For the discrete variables with nominal values or Boolean (yes/no) – the values: 0, 1, 2…… n-1 was assigned to each value for n labels; this is the case for Genotype as SS=0, SE=1 and S*=2. For the continuous variables which are measured; a value of the percentage expressed as a proportion of 1 was used, i.e. 10% and 56% read as 0.10 and 0.56 respectively into the appropriate membership functions. Table 1 gives a description for the values of the labels to be used for each variable along with their respective membership functions.

Table 1: Description of the labels for each variables

№	Variables	Membership function mapping (label = value)			Membership function
1	Level of Fetal Haemoglobin	<=2% becomes <=0.02 $\Rightarrow x_1 \leq 0.02$	2%<level<=5% becomes 0.02<level<=0.05 $\Rightarrow 0.02 < x_1 \leq 0.05$	>5% becomes >0.05 $\Rightarrow x_1 > 0.05$	Trapmf
2	Type of Genotype	SS $\Rightarrow x_2 = 0$	SE = 1 $\Rightarrow x_2 = 1$	S* = 2 $\Rightarrow x_2 = 2$	Trimf
3	Degree of Anemia	<15% becomes <0.15 $\Rightarrow x_3 < 0.15$	>=15% becomes >=0.15 $\Rightarrow x_3 \geq 0.15$		Trapmf
4	Likelihood of SCD	No $\Rightarrow y = 0$	Yes $\Rightarrow y = 1$		Trimf

3.5 Fuzzification of the variables

For the purpose of this study, the triangular and trapezoidal membership functions were used to map the degree of membership of the labels of each variable used both input and output variable. Following is a description of each variable and the type of membership function used for the labels alongside the ordered pair that was used in mapping the degree of membership for each variable's label.

a. Level of Fetal Haemoglobin, x1 (Figure 3)

- $x_1 \leq 0.02$ - trapmf[x1; -0.36, -0.004, 0.016, 0.02]

$$\mathrm{trapmf}[x1;\ -0.36, -0.004, 0.016, 0.02] = \left\{\begin{array}{cc} 0, & x1 \leq -0.36 \\ \dfrac{x1 + 0.36}{0.356}, & -0.36 \leq x1 \leq -0.004 \\ 1, & -0.004 \leq x1 \leq 0.016 \\ \dfrac{0.02 - x1}{0.004}, & 0.016 \leq x1 \leq 0.02 \\ 0, & 0.02 \leq x1 \end{array}\right\} \quad (2)$$

- $0.02 < x_1 \leq 0.05$ - trapmf[x1; 0.0186, 0.0202, 0.0376, 0.0476]

$$\mathrm{trapmf}[x1;\ 0.0186, 0.0202, 0.0376, 0.0476] = \left\{\begin{array}{cc} 0, & x1 \leq 0.0186 \\ \dfrac{x1 - 0.0186}{0.0016}, & 0.0186 \leq x1 \leq 0.0202 \\ 1, & 0.0202 \leq x1 \leq 0.0376 \\ \dfrac{0.0476 - x1}{0.01}, & 0.0376 \leq x1 \leq 0.0476 \\ 0, & 0.0476 \leq x1 \end{array}\right\} \quad (3)$$

- $x_1 > 0.05$ - trapmf[x1; 0.05, 0.062, 1.142, 1.462]

$$\text{trapmf}[x1;\ 0.05, 0.062, 1.142, 1.462] = \begin{cases} 0, & x1 \le 0.05 \\ \dfrac{x1 - 0.05}{0.012}, & 0.05 \le x1 \le 0.062 \\ 1, & 0.062 \le x1 \le 1.142 \\ \dfrac{1.462 - x1}{0.32}, & 1.142 \le x1 \le 1.462 \\ 0, & 1.462 \le x1 \end{cases} \tag{4}$$

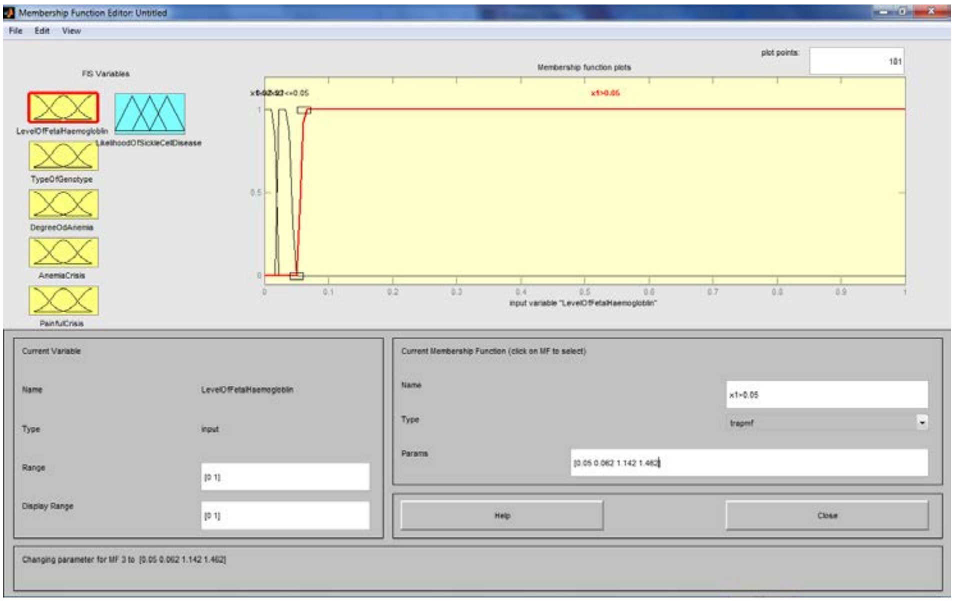

Figure 3: Membership function for the Level of Fetal Haemoglobin

b. Type of genotype, x2 (Figure 4)

- SS; $x_2 = 0$ - trimf[x2; -0.5, 0, 0.44]

$$\text{trimf}[x2;\ -0.5, 0, 0.44] = \begin{cases} 0, & x2 \le -0.15 \\ \dfrac{x2 + 0.5}{0.5}, & -0.15 \le x2 \le 0 \\ \dfrac{0.44 - x2}{0.44}, & 0 \le x2 \le 0.44 \\ 0, & 0.44 \le x2 \end{cases} \tag{5}$$

- SE; $x_2 = 1$ - trimf[x2; 0.5, 1, 1.44]

$$\text{trimf}[x2;\ 0.5, 1, 1.44]$$

$$= \begin{cases} 0, & x2 \le -0.15 \\ \dfrac{x2 - 0.5}{0.5}, & -0.15 \le x2 \le 1 \\ \dfrac{1.44 - x2}{0.44}, & 1 \le x2 \le 1.44 \\ 0, & 1.44 \le x2 \end{cases} \tag{6}$$

- S*; $x_2 = 2$ - trimf[x2; 1.5, 2, 2.44]

$$\mathrm{trimf}[x2;\ 1.5, 2, 2.44] = \begin{cases} 0, & x2 \le 1.15 \\ \dfrac{x2 - 1.5}{0.5}, & 1.15 \le x2 \le 2 \\ \dfrac{2.44 - x2}{0.44}, & 2 \le x2 \le 2.44 \\ 0, & 2.44 \le x2 \end{cases} \tag{7}$$

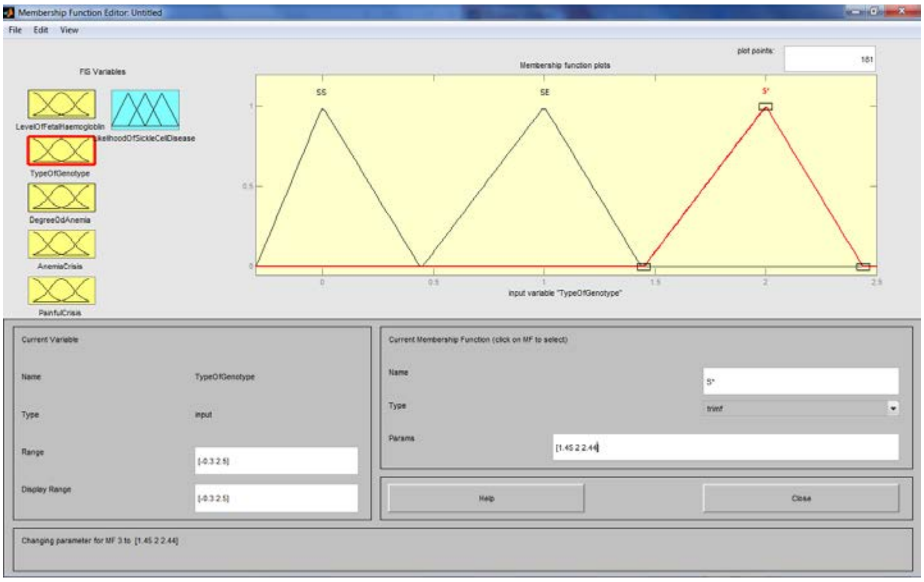

Figure 4: Membership function for the type of genotype

c. Degree of anemia, x3 (Figure 5)

- $x_3 < 0.15$ - trapmf[x3; [0, 0.001, 0.011, 0.0144]

$\mathrm{trapmf}[x3;\ [0, 0.001, 0.011, 0.0144]$

$$= \begin{cases} 0, & x3 \le 0 \\ \dfrac{x3}{0.001}, & 0 \le x3 \le 0.001 \\ 1, & 0.001 \le x3 \le 0.011 \\ \dfrac{0.011 - x3}{0.0034}, & 0.011 \le x3 \le 0.0144 \\ 0, & 0.0144 \le x3 \end{cases} \tag{8}$$

- $x_3 \ge 0.15$ - trapmf[x3; 0.015, 0.02, 0.99, 1]

$\mathrm{trapmf}[x3;\ 0.015, 0.02, 0.99, 1]$

$$= \begin{cases} 0, & x3 \le 0.015 \\ \dfrac{x3 - 0.015}{0.005}, & 0.015 \le x3 \le 0.02 \\ 1, & 0.02 \le x3 \le 0.99 \\ \dfrac{0.99 - x3}{0.01}, & 0.99 \le x3 \le 1 \\ 0, & 1 \le x3 \end{cases} \tag{9}$$

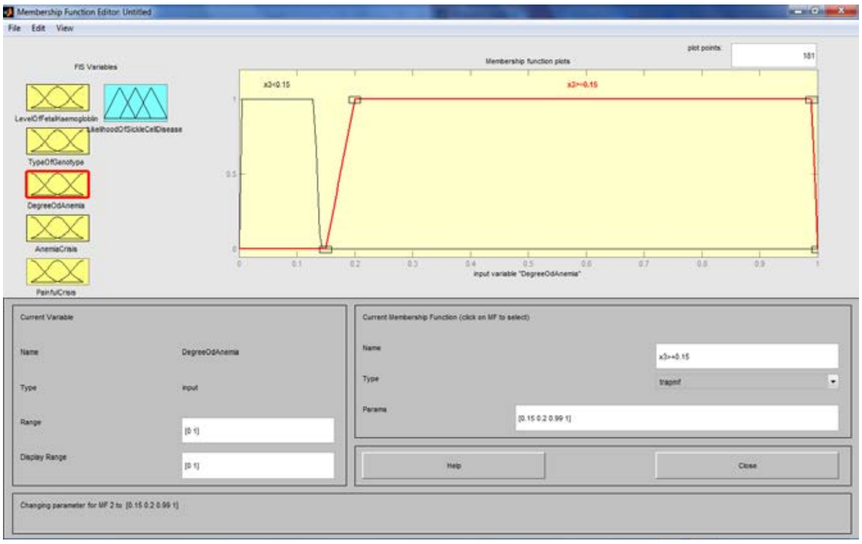

Figure 5: Membership function for the degree of Anemia

d. Likelihood of Sickle-cell disease, y (Figure 6)

- No; y = 0 - trimf[y; -0.5, 0, 0.44]

$$\text{trimf}[\text{y};\ -0.5, 0, 0.44] = \begin{cases} 0, & y \le -0.5 \\ \dfrac{y+0.5}{0.5}, & -0.5 \le y \le 0 \\ \dfrac{0.44-y}{0.44}, & 0 \le y \le 0.44 \\ 0, & 0.44 \le y \end{cases} \tag{10}$$

- Likely; y = 1 - trimf[y; 0.5, 1, 1.44]

$$\text{trimf}[\text{y};\ 0.5, 1, 1.44] = \begin{cases} 0, & y \le 0.5 \\ \dfrac{y-0.5}{0.5}, & 0.5 \le y \le 1 \\ \dfrac{1.44-y}{0.44}, & 1 \le y \le 1.44 \\ 0, & 1.44 \le y \end{cases} \tag{11}$$

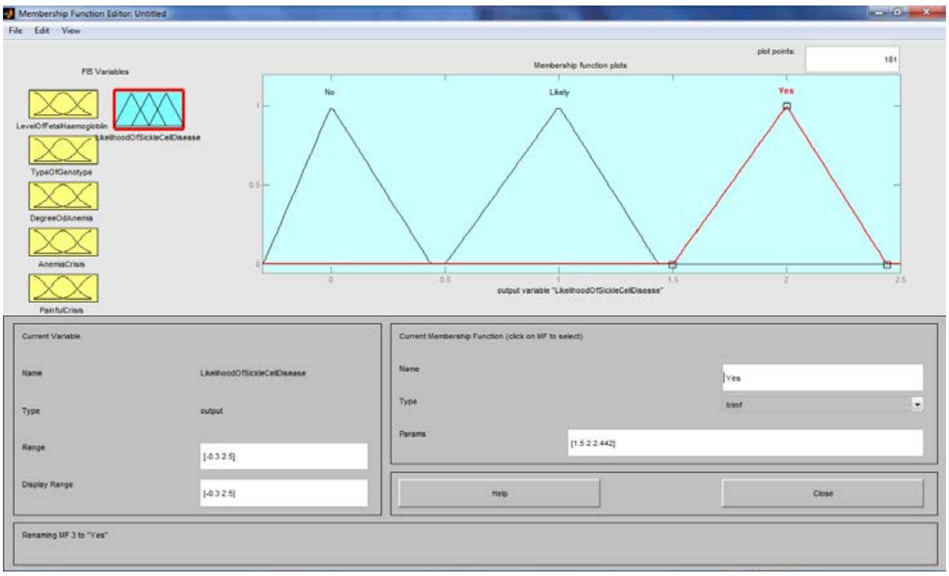

Figure 6: Membership function for the degree of Anemia

3.6 Inference engine development, aggregation and defuzzification

After developing the membership function, the process of developing the fuzzy inference engine which makes use of the 18 different rules shown in Table 2 below is necessary. It is with the information about the membership functions i.e. the labels that have been used to map each interval of membership functions e.g. Genotype had labels: SS, SE and S* that the rules were been formed.

Hence, each rules that was provided is a result of a case-based reasoning approach which involves the experience that the expert had had in the years have shown such pattern except otherwise cases where there were misdiagnosis (false positives) or undiagnosis (cases not yet understood).

For the purpose of this study and the variables that are considered – the And Method used in evaluating each degree of membership is Minimum (it selects the smallest value of many), the Or Method used is the Maximum (it selects the largest value out of many); which although is not used in this study and the Implication Method used is the Minimum. These fuzzy operators were used to calculate the output for each rules which now require aggregation to be applied in order to get a single output.

The Aggregation method used in determining the optimum output membership function for the output is chosen to be Maximum (it selects the largest value for every region of the output variable's membership function). This method was chosen since it is the most commonly used method of aggregating linear-wise membership functions like trapezoidal and triangular membership functions.

Table 2: Fuzzy rules developed for the inference system

№	Hemoglobin level	Genotype	Degree of Anemia	Sickle Cell Survival (Yes/No)
1	<=2%	SS	<15%	No
2	<=2%	SS	>=15%	No
3	<=2%	SE	<15%	No
4	<=2%	SE	>=15%	Yes
5	<=2%	S*	<15%	No
6	<=2%	S*	>=15%	Yes
7	2%<level<=5%	SS	<15%	No
8	2%<level<=5%	SS	>=15%	Yes
9	2%<level<=5%	SE	<15%	Yes
10	2%<level<=5%	SE	>=15%	Yes
11	2%<level<=5%	S*	<15%	Yes
12	2%<level<=5%	S*	>=15%	Yes
13	>5%	SS	<15%	No
14	>5%	SS	>=15%	Yes
15	>5%	SE	<15%	Yes
16	>5%	SE	>=15%	Yes
17	>5%	S*	<15%	Yes
18	>5%	S*	>=15%	Yes

The defuzzification of the output membership function resulting from the process of aggregation shows the crisp result that gives the likelihood of sickle cell disease as a real number value (a value within the range of the output variable's membership functions). In the case of this study the values 0 (or between 0 and 0.44) and 1 (or between 0.5 and 1.44) were used to identify No and Yes respectively. The method of defuzzification chosen for this study is the centroid method – it simply calculates the centre-of-gravity of the final polygon that results from the process of aggregation. It is also chosen for its compatibility with linear-wise membership functions.

Figure 7 shows the diagram of the simulated fuzzy logic system for the prediction of the likelihood of Sickle-cell disease in an individual given the values for three (3) input variables, namely: level of fetal haemoglobin, type of genotype and degree of anemia. This is the view of the fuzzy inference system using the fuzzy logic toolbox available in the MATLAB R2012a software used as the simulation environment.

The simulated fuzzy logic system is of the Mamdani type and consists of 3 and 1 fuzzified inputs and output variables and an inference engine which contains 18 IF-THEN rules describing expert knowledge - gathered via knowledge elicitation from expert.

This fuzzy logic system can be used to determine what the likelihood of sickle-cell disease will be; given a known set of clinical data on individuals which contain the under-listed variables. This will be used as a means of evaluating the accuracy of the system by putting into consideration the true and false positive outcomes.

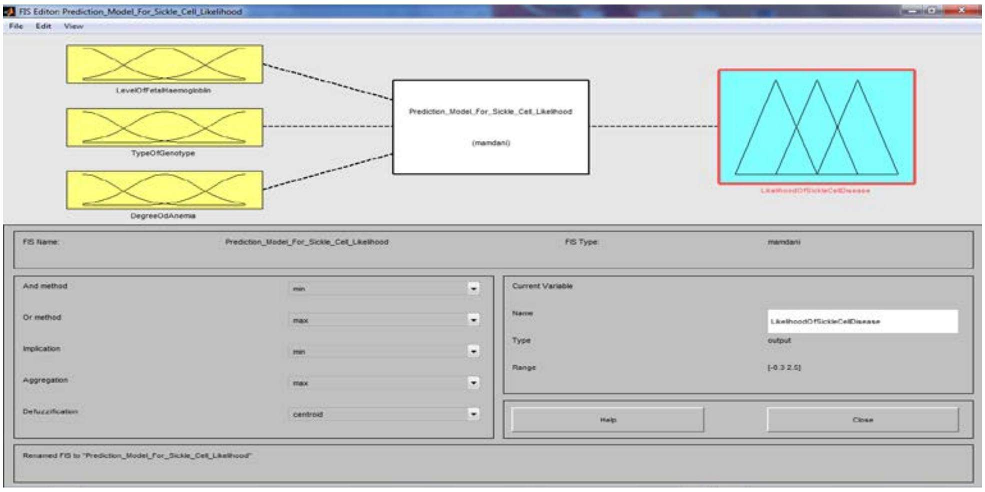

Figure 7: Proposed fuzzy inference system

4 Results and Discussions

After formulating the model necessary for simulating the fuzzy logic inference system – the model was implemented using the MATLAB Versions 7 software developed as Release 2012. The fuzzy logic toolbox which is among the many toolboxes available in the MATLAB software was used in simulating the predictive model using triangular and trapezoidal membership functions for the fuzzification of the input and output variables. The fuzzy logic system was used to perform a view of the surface diagram which shows the distribution of the many possible values and the relationship between any two variables.

Figure 8 gives a plot of the surface diagram showing the relationship between level of haemoglobin and the type of genotype; it can be observed that the diagram clearly shows that there is more likelihood of cases of people having haemoglobin levels above 2% to have SCA while cases having haemoglobin levels below 2% are more likely not to have SCD. The plot also shows that for all cases of fetal haemoglobin levels with and SS genotype there is hardly any likelihood of the sickle-cell disease occurring but for cases where the genotype is either SE and S* the patients have a likelihood of SCD.

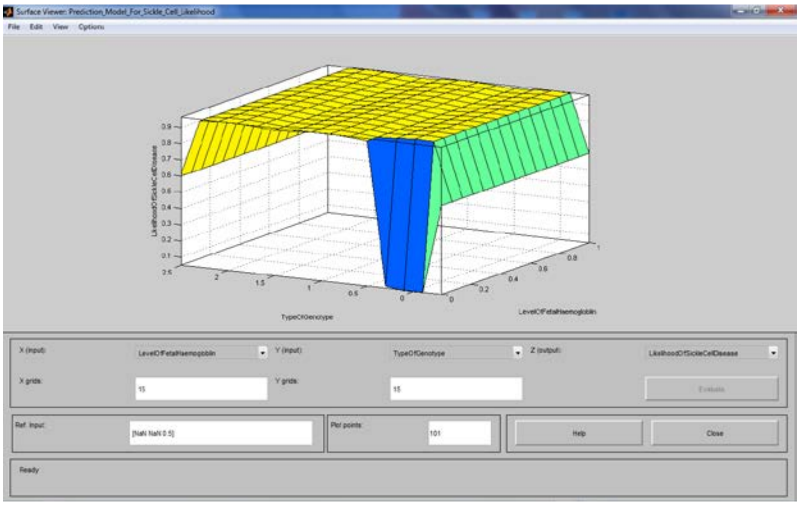

Figure 8: Surface diagram of fetal haemoglobin and genotype

Figure 9 shows the surface diagram having a plot of the relationship between the level of haemoglobin and the degree of anemia. It can be observed that for cases where the degree of anemia is less than 15% there is no likelihood of having SCD. And for cases where the level of fetal haemoglobin is less than or equal to 2% there is hardly the possibility of having SCA except where the genotype is SE or S*.

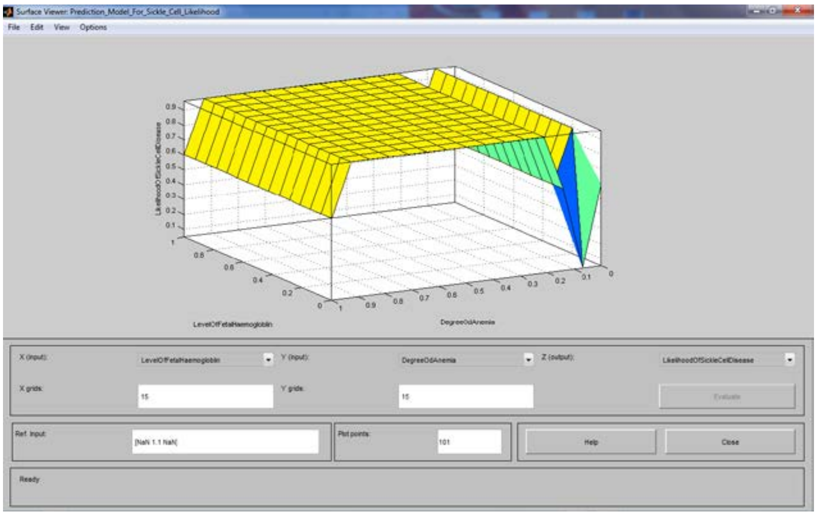

Figure 9: Surface diagram of the level of fetal haemoglobin and genotype

Figure 10 shows the surface diagram of the relationship that exists between the data values of the degree of anemia and the type of genotype of the patient. It was observed that whenever the genotype is SS there is no likelihood of SCA but when the level of fetal haemoglobin is less than or equal to 2% there is the likelihood of SCD. But for all cases where the degree of anemia is greater than or equal to 15% there is a likelihood of the existence of the SCD.

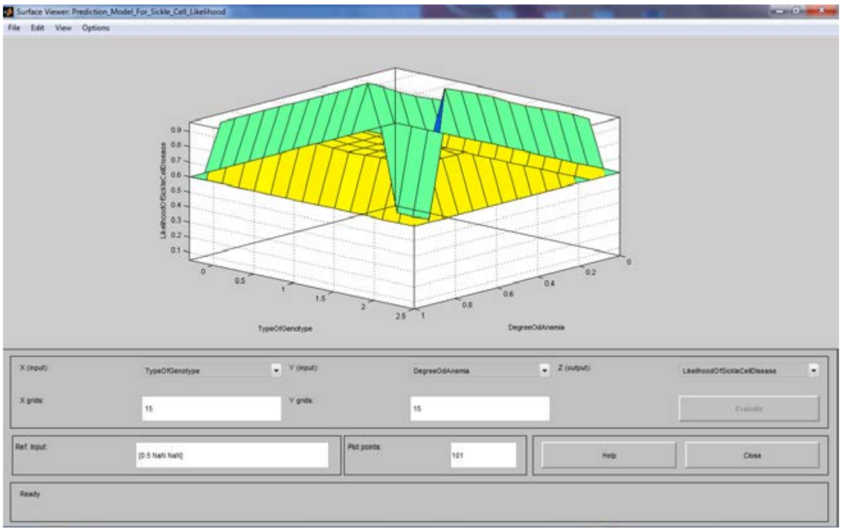

Figure 10: Surface diagram degree of anemia and genotype

5 Conclusion

The proposed model for the prediction of the likelihood of SCA was presented using 3 input variables namely: the level of fetal haemoglobin, degree of anemia and the type of genotype of the patient. The variables were identified and knowledge defining the relationship between variables was used in developing the inference system of the fuzzy inference system. The variables were all fuzzified and the fuzzified input variables were fed to the inference engine. The 18 output that were produced after the inference engine are aggregated to a single output which was defuzzified to get the crisp output i.e. No or Yes.

The model was simulated using the fuzzy logic toolbox available in the MATLAB software and the results of the behavior of the proposed model presented via the surface diagram. It is believed that this model will help diagnose the likelihood of SCA in an individual having provided a record containing the inputs as a 3-tuple. This model should help reduce the number of untimely deaths which occur as a result of late detection.

REFERENCES

[1]. Malowany, J. I., Butany, J. (2012). Pathology of sickle cell disease. *Seminars in Diagnostic Pathology*, 29(1), pg. 49-55.

[2]. Mishra, N., Jha, P. (2014) Framing Fuzzy Rules using Mamadani Model for Effective Sickle Cell Diagnosis. *International Journal of Engineering Sciences and Research Technology*. 3(6), June 2014, pg. 389-396, ISSN: 2277-9655.

[3]. Obeagu, E. I., Ogbuabor, B. N., Ikechukwu, O. A., Chude, C. N. (2014) Haematological parameters among sickle cell anemia patients' state and haemoglobin genotype AA individuals at Michael Okpara University of Agriculture, Umudike, Abia State, Nigeria. *International Journal of Current Microbiology and Applied Sciences*; 3(3), pg. 1000 – 1005, ISSN: 2319-7706.

[4]. Omoti, C. E.2005.Haematological Values in Sickle Cell Anaemia in Steady State and During Vaso-occlusive Crisis in Benin City, *Nigeria Annals of African Medicine*, 42, pg.62-67.

[5]. Pauling, L., Itano, H.A., Singer, S. J., Wells, I. C. (1949). Sickle Cell Anaemia: a Molecular Disease. *Journal of Science* 110, pg. 543-548.

[6]. Ukpong, E. (1992) Current Concepts in the Management of Sickle Cell Disorders. Kraft Books, Ibadan.

[7]. Wellems, T. E., Hayton, K., Fairhurst, R. M. (2009). The impact of malaria parasitism: from corpuscles to communities. *Journal of Clinical Investment*; 119 (9), pg. 496-505.

[8]. Aramideh, J., Jelodar, H. (2014) Application of Fuzzy logic for presentation of an expert system to diagnose anemia. *Indian Journal of Science and Technology,* 7(7), pg. 933-938, July 2014. ISSN: 0974-6846.

[9]. Sayyahi, H. (2008). An application of fuzzy case based reasoning for poison classification. Unpublished Masters in Information Technology Thesis. University of Malaysia.

[10]. Goyal, D. (2006) Disease Diagnostic System using LabVIEW. Unpublished Masters of Engineering in Electronics Instrumentation and Control Engineering Thesis. Thapar Institute of Engineering and Technology, Deemed University, Punjab.

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Network-aware Composition for Internet of Thing Services

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ABSTRACT

To enhance the adoption of Internet of Things (IoT) philosophy for the internet, research into IoT service composition has gathered momentum. In a distributed IoT environment, identifying IoT service among a set of similar service offerings that meets both functional and performance requirements of an IoT application has become important. However, the performance of a service cannot be guaranteed. Therefore service's QoS and network characteristics are required to aggregate IoT services. Most existing composition approaches only consider non-network related QoS properties at the application tier. However they do not consider the network parameters such as network latency at the application level in selection and composition of services. Therefore we propose two evolutionary algorithms for IoT service composition that consider not only QoS but also network latency at the IoT application layer. The algorithms are discussed and results of evaluation are presented. The results indicate that our algorithms are efficient in finding QoS optimal and low latency solutions.

Keywords: Internet of Things, Service Composition, QoS, Network latency, Composite service, Evolutionary Algorithm

1 Introduction

Internet of Things (IoT) envisions the internet as a set of interconnected objects. Objects refer to physical smart devices that utilize computing resources such as CPU, memory and network capabilities [1] in exposing their functionalities via the internet to the outside world. Recently, research studies [10][8][12][9] have attempted to map IoT as a service-oriented framework where smart device functionalities are exposed as services. This allows for the development of loosely coupled IoT applications [2] in which services can be discovered and selected according to user requirement. Given the immense potential of applying service-oriented concepts to IoT domain, several challenges have been identified. When single service is not sufficient in meeting user requirements, services will need to be combined into composite service that provide more complex capabilities that meet user needs. IoT service selection usually involves comparing service QoS scores. QoS represents the non-functional aspects of a service such as price, availability, reputation, response time, etc. It serves as a criteria that differentiates services offering similar capability. Once a service is selected based on functionality, composition can be carried out to find the right combination of services that yield composite service with optimal QoS. As IoT applications are becoming more distributed over the internet, network latency has become important in determining the performance of composite services. This is especially evident in IoT applications that require some form of data streaming. Data streaming is one of the most significant requirement of an IoT network [1] and is heavily dependent on network latency. Network latency, otherwise known as round trip time (RTT), is defined as the amount of time required for network packets to take a round

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trip from a source node to destination node [17]. For instance a data-intensive IoT application is presented in Figure 1. The purpose of the application is to provide video or audio feeds from a variety of smart devices like an internet-enabled surveillance camera or a Wifi-enabled push-to-talk (PTT) phone.

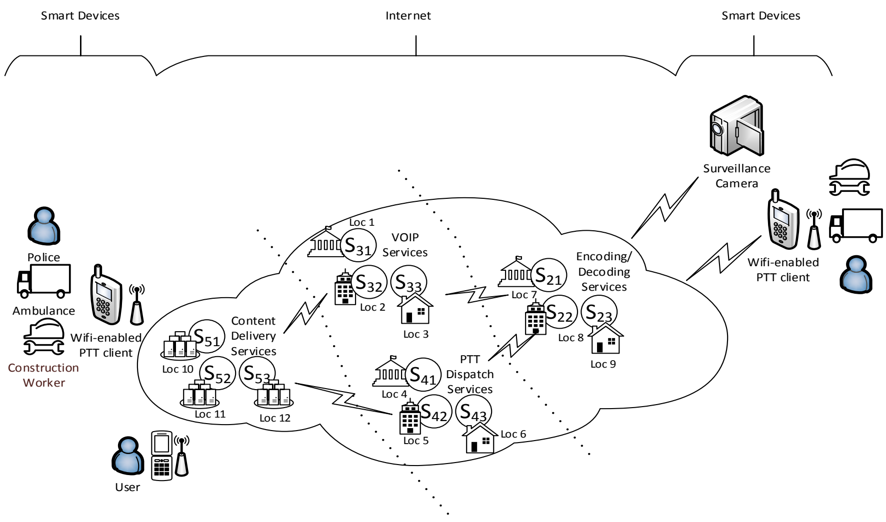

Figure 1: IoT services application

The application is composed of a variety of services deployed in different network locations on the internet and are participants of a composition process in order to provide much more advance features for the user. A typical scenario is when a police officer wants to view a video stream coming from a surveillance camera in a remote location on his mobile phone via a Wifi connection. As there exists many processes involved such as encoding/decoding, VoIP streaming and content delivery, such a scenario would involve the orchestration of different services that can perform each process. For example, alternative services (candidate services) like S31, S32 and S33 in Figure 1 can take care of VoIP streaming, while services S51, S52 and S53 representing different cloud-based content delivery networks (CDN) can handle content delivery to the police officer's smart phone. A simple composite services could be formed from integrating any of Encoding services with VoIP and CDN services. The problem now becomes how to integrate one service from each set of alternative services into a composite service in such a way that it satisfies the user's QoS and network requirement. By network requirement, we mean composition process should take into consideration the services that are closer to each other in terms of their network locations (as represented by inter-service RTT values). This will ensure optimum network performance of the application from the user's perspective.

In order to measure inter-service network latency, state of the art measurement tools could be deployed at the network layer. These tools function by measuring packet pings between all service nodes within a composition in order to obtain their network locations. The network locations are then fed to the application layer where the service composition process tries to obtain the end-to-end network latency for the composite service. Due to congestion and packet collisions, the tools are very slow in measuring RTT and therefore are not useful in situations where the IoT application requires data streaming or has strict time constraints such in Figure 1. As such, rather than measuring RTT, our proposed techniques estimate network latency QoS at the application level (i.e. during service composition process) so as to search for composite service that meets not only QoS constraints, but also has near-optimal end-to-end network latency of its execution path. In other words, the optimal solution should have the best balance between optimal QoS score and network latency.

The problem of service composition has been described as NP-Hard Problem [18]. This is so because as the number of alternative services on the internet has increased, leading to a rise in number of possible composite services. This will also cause an exponential increase in the time it takes to find an optimal solution. To facilitate composition of IoT services which can be a time consuming process [15], research efforts have developed several algorithms capable of aggregating IoT services that contribute to optimum composite service and meets users' QoS constraints. [19] Tackle service composition in very large-scale IoT systems. They present an architecture that adapts composition process depending on the availability of constituent smart devices offering medical services such as ambulance, health insurance etc. The architecture considers the choreographic aspects of service composition. However it does not consider the QoS or network aspect of service composition. Another study [20] develop two probabilistic model for IoT service composition. The first model is based on finite state machine and deals with only the functional aspect of composition process while the second model is based on Markov Decision Process that handles service cost and reliability QoS properties. The study also falls short of considering Network-centric QoS. The most popular choice of algorithms for tackling QoS optimization of services are Evolutionary algorithms. Evolutionary algorithms are techniques that operate on the concepts of natural evolution [5]. They have shown great promise in tackling service composition problem because they add characteristics such as population diversity, reinforced learning, memory and adaptability to the composition process. They have also been shown to be more computationally efficient than other types of algorithms. Several evolutionary service composition techniques such as in [11] [13] [22] have been developed, although very few have been applied to the domain of IoT services. One such studies is in [21] which introduce a hybrid cooperative evolution algorithm that combines elements of Genetic and Particle Swarm Algorithms in searching for QoS-optimal composite services. The result is an algorithm that adapts to real-time data streaming from services running on smart devices. A similar study in [14] present a multi-objective approach to QoS-based service composition using a Genetic algorithm. Both approaches take service QoS into consideration, but once again ignore the network aspect of QoS.

Conclusively, recent works demonstrate good capability in finding non network-centric QoS optimal solutions, however they fail to consider the impact of network-related QoS parameter such as network latency on service composition at the application level. In contrast, we propose two evolutionary approaches to network-aware IoT service composition. Our approaches utilize a QoS model that is extended with a network model which efficiently estimates the RTT between services running on IoT devices. The network model consists of a decentralized network coordinate system for fast estimation of end-to-end RTT. This will ensure that RTT estimation process does not negatively impact the overall computation time for our approach and for the IoT application. We also propose novel network-aware Genetic and Particle swarm algorithms for searching compositions with both optimal QoS and optimal network latency. We then compare our approaches against each other and against other state of the art techniques and present the results of our experiment. The remainder of the paper is organized as follows. Section 2 formulates the service composition problem. Section 3 presents our network model and proposed techniques. Section 4 discusses the result of evaluation of our approaches. Section 5 concludes this paper.

2 Problem Formulation

Service composition borrows its concepts from workflow management systems [3] where a functionality is implemented as a task, and process of task execution follows specific workflow pattern which could be one of the many patterns such as sequence, parallel, loop, etc. The goal of service composition is to find a set of interconnected services (one per task) that contribute to the optimal composite service and meets user's need. With this in mind, the service composition problem is described as follows:

Given a set of n interconnected tasks that are needed to satisfy a user requirement,

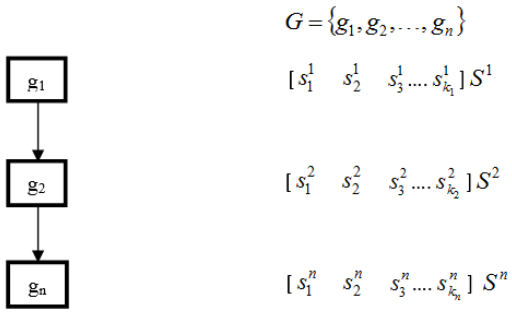

Figure 2: Arrangement of candidate services into tasks

Every task requires *k* number of similar services or candidate services that have the ability to complete the task,

$$S^i = \{s_1^i, s_2^i, \ldots, s_k^i\},\ \forall i \in [1..n]$$

Where i identifies the class in which similar services are grouped according to their task g_i as seen in Figure 2. Only one candidate service s_j^i is selected and bound to its task. Once all tasks have been bound a composite service C is formed,

$$C = \{s_j^1, s_j^2, \ldots, s_j^n\},\ \forall j \in [1..k]$$

Every service is assigned a number QoS values. In our work we consider four QoS objectives namely cost, response time, reputation and network latency. For a composite services, end-to-end QoS values are computed by aggregating the QoS values of constituent services depending of type of workflow. End-to-end cost, response time, reputation and network latency are calculated using (1) (2) (3) and (4) respectively;

$$Q_X(C) = \sum_{i=1}^{n} X(s_j^i) \qquad (1)$$

$$Q_R(C) = \sum_{i=1}^{n} R(s_j^i) \qquad (2)$$

$$Q_P(C) = \frac{\sum_{i=1}^{n} P(s_j^i)}{n} \qquad (3)$$

$$Q_L(C) = \sum_{i=1}^{n} L_{s_j^{i+1}}^{s_j^i} \qquad (4)$$

Where $L_{s_j^{i+1}}^{s_j^i}$ represents the round trip time (RTT) between candidate services S_j^i and S_j^{i+1}. Q_X, Q_R, Q_P and Q_L represent end-to-end QoS value for cost, response time, reputation and network latency. Also X, R, P and L represent service's QoS value for each QoS objective respectively. While

Normalization of cost, response time and reputation in the range [0 1] is achieved to obtain a fitness value (*F*) using (5). Where $Max_m(S^i)$ and $Min_m(S^i)$ represent maximum and minimum QoS values for service class *i.*

$$F_m(C) = \sum_{\substack{i=1 \\ m\in\langle X,R,P\rangle}}^{n} \left(\frac{Max_m(S^i) - Q_m(S_j^i)}{Max_m(S^i) - Min_m(S^i)} \right) \tag{5}$$

While fitness value for network latency is obtained using (6),

$$F_L(C) = \frac{Q_L(C)}{H} \tag{6}$$

Where H is a constant which normalizes value of $Q_L(C)$ in the range of [0 1].

The service composition problem becomes a multi-objective optimization problem where the aim is to search for composite services with optimal fitness values with respect to cost, response time, reputation and network latency, subject to the following constraints:

- One service should be selected for each task
- QoS boundary constraint:

$$\forall Q_X \in \left[q_X^{\min} \quad q_X^{\max} \right], \ \forall Q_R \in \left[q_R^{\min} \quad q_R^{\max} \right], \ \forall Q_P \in \left[q_P^{\min} \quad q_P^{\max} \right]$$

3 Network Model

In this work we adopt a network model that efficiently estimates network latency between service nodes in a network. The model consists of a network coordinate system [4] that computes round trip times (RTT) between service nodes. Ordinarily, RTT values are measured by sending network packets across the network and obtaining the time it takes them to reach their destination. Unfortunately this approach is not scalable and will cause computation overhead on the network. In comparison, our network coordinate system works by only measuring RTT from each service node to a small subset of neighbors. The measurements are then used to estimate un-measured RTT to other nodes. Here we adopt state of the art network coordinate system based on Matrix factorization [4]. Once RTT between all service nodes have been determined, the values are fed to our novel service composition approaches to establish network-awareness during composition process. The algorithm for network coordinate system is outlined in Figure 3.

```
1. Initialize environment parameters: M, Z, maxiter; Where Z is total
   number of service nodes and maxiter is maximum iteration no.
2. Measure RTT between each service node and a small number
   M of random neighbours;
3. For each node select (Z-M) neighbours with un-measured RTT;
4. Initialize dataset D with both measured and un-measured RTT.
5.  For i = 1 to maxiter
6.        U = rand (position_1); V = rand (position_2);
7.        X = U * V^T;
8.        error = w(D – X)^2;
9.        If (error is not minimized)
10.              Return;
11.       End If;
12. End For;
12. Return
```

Figure 3: Outline of Network latency estimation algorithm

4 Evolutionary Algorithms for Network-Aware Service Composition

4.1 Evolutionary Particle Swarm Algorithm

Our first proposed approach is an Evolutionary Particle Swarm algorithm. Classic Particle Swarm optimization algorithm (PSO) [6] carries out optimization by encoding the problem using swarms of particles that iterate their velocity and position attributes until an optimal solution is found. However it is plagued with premature convergence, poor swarm diversity and lack of alternative optimal solutions. In order to avoid these problems, we adapt classic PSO with evolutionary concepts like multi-population and non-dominated sort ability in performing optimization. The resultant algorithm is called Evolutionary Particle Swarm or VPSO. It aims to search for a Pareto set of composite services that have optimal QoS.

Encoding

The algorithm encodes composite service as a particle array where each array element (e1, e2…en) represents a task that can be bound to any candidate service.

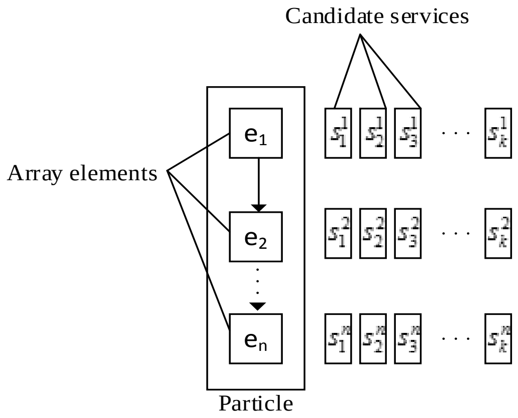

Figure 4: Encoding of a particle

Population initialization

It starts optimization process by initializing population of particles referred to as D population. This is achieved by arbitrarily selecting one candidate service for each task until all particles are initialized.

With the aid of RTT values earlier estimated by the network coordinate system, every particle is then assigned QoS values and a starting velocity and position.

Non-dominated Sorting and Multi-population creation

In the next step, the population is passed through a non-dominated sort process which involves sorting particles according to their fitness values and assigning fronts to each particle based on degree at which it dominates other individuals. For example, particle T_a dominates another particle T_b if its fitness value in all QoS objectives (cost, response time, reputation and network latency) are better than fitness values of T_b. Therefore T_a will be placed in a higher front than T_b. Once particles have been sorted, the top 25% of the population is placed in a second population called O population. Then crowding distance (CD) value is computed for each individual. This value determines the Euclidean distance between a particle and its neighbors. CD is an important value because it helps the algorithm to determine diversity or spread between individuals in the O population. The next stage involves creation of a third population known as N population consisting of individuals from D population with the best network latency value.

Updating Particle Velocity and Position

With the aid of first, second and third populations, new values for particle velocity and position are calculated with (7) and (8),

$$V^i_{j(new)} = wV^i_j + c_1r_1(N^i_j - D^i_j) + c_2r_2(O^i_j - D^i_j) \tag{7}$$

$$D^i_{j(new)} = D^i_j + V^i_{j(new)} \tag{8}$$

Where *V* is particle velocity; *w* is inertia weight, c_1 and c_2 represent constants; r_1 and r_2 are random numbers in range [0 1]; *N* is population of particles with best cost, response time and reputation; *O* denotes population of particles with best network latency; *D* represents initial population. Equations (7) and (8) force particle towards areas in search space where QoS objectives have good fitness values in terms of both QoS and Network latency as defined by their velocity. Where a particle's velocity is directly proportional to both the distance between the particle and particle with best network latency (i.e. $(N^i_j - D^i_j)$), and the distance between the particle and particle with best QoS score (i.e. $(O^i_j - D^i_j)$). Typically particles with lower velocities will move more slower than particles with higher velocities in the search space thereby keeping best particles (with low velocity) for participation in subsequent populations, while bad particles (with high velocity) are changed to new individuals. The effectiveness of equations (7) and (8) are demonstrated by result of experiment in the next section. VPSO is summarized in Figure 5.

```
1. Initialize PSO parameters: gen, pop_size, c1, c2
2. D = generate_population(gen, pop_size);
3.  O = non_dominated_sort(D);
4. For i = 1 to gen
5.          N = sort(O);
6.          For i = 1 to pop_size
7.                    V(new) = wV + c1r1(N – D)+c2r2(O – D);
8.                    D(new) = D + V(new);
9.                    D = compute_qos(D);
10.         End For
11.         O = non_dominated_sort(D);
12. End For
13. Return
```

Figure 5: Outline of VPSO algorithm

4.2 Network-aware Genetic Algorithm

In our second approach, we develop a novel Genetic algorithm based on non-dominated sort called N-Genetic algorithm or NGA. Non-dominated sort Genetic algorithms [16] are a class of Genetic algorithms that are capable of tackling problems of a multi-objective nature. They are also able to perform non-dominated sort operation in addition to standard operations such as crossover and mutation.

Encoding

NGA encodes composite service as a genome which consist of genes instead of elements as in the case of VPSO. A gene represents a candidate service bound to its task and can take an integer value.

Population Initialization

Similar to VPSO, NGA initiates optimization process by creating an initial population of individuals with initialized QoS values. Non-dominated sort operation is then performed on the initial population to find individuals with the best fitness values.

Crossover Operation

The individuals (parents) are subsequently placed in a mating pool and then subject to crossover operation. The operation involves intertwining sets of genes between any two parents. The type of operator used in our work is a single point crossover operator.

Mutation Operation

In order to integrate network awareness into NGA, we run k-means clustering algorithm [7] over our network model in order to effectively classify service nodes into separate clusters according to their RTT distance from other nodes. This way, services that are closer together in RTT are placed in the same cluster, while services that are further away are placed in different clusters. The clusters are then used by mutation operator to determine which genes within the same cluster that will be candidates for replacing the gene to be mutated. The operator will arbitrarily select only one gene among candidates available as seen in Figure 6. NGA algorithm is described in Figure 7.

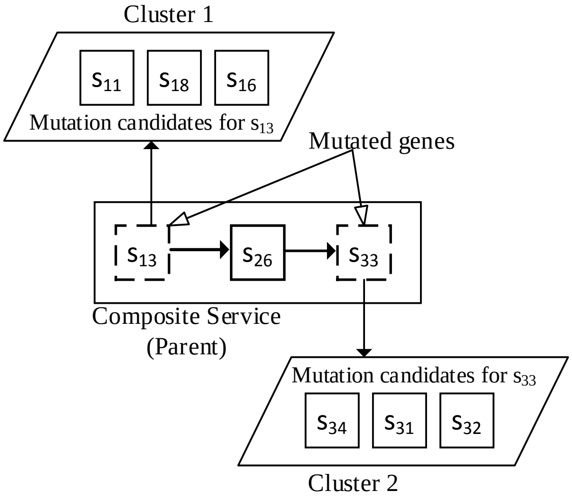

Figure 6: Mutation Operation of NGA

```
1. Initialize GA paramete: : gen, pop_size.
2. pop = generate_population(gen, pop_size);
3.  pop = non_dominated_sort(pop);
4. For i = 1 to gen
5.          parent_pop = tournament_selection(pop);
6.          parent_pop = single_crossover_operation(parent_pop);
 7.         parent_ pop = non_dominated_sort(parent_pop);
8.          child_pop = mutation_operation(parent_pop);
9.          combination_pop = pop + child_pop;
10.         combination_pop = non_dominated_sort(combination_pop);
11.         pop = replacement (combination_pop);
12. End For
13. Return
```

Figure 7: Outline of NGA algorithm

5 Experiments and Analysis

In order the test the proposed algorithms VPSO and NGA, we present in Figure 8 a set of sequence workflows from our IoT application scenario in Figure 1 for sake of simplicity. It is expected that the results obtained will be similar irrespective of the sequence workflow used.

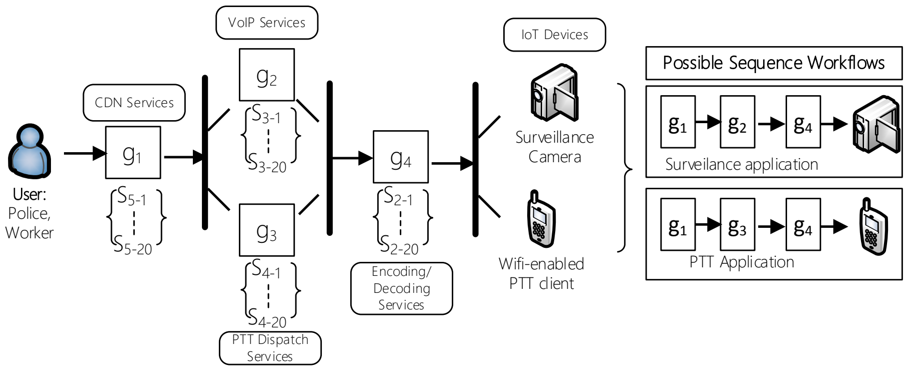

Figure 8: Experimental Service Composition Scenario

The experiment was executed in MATLAB 2013 running on Intel Core i7 (3.6GHz) CPU with 8GB RAM memory. Part of our experiment will aim to investigate how our algorithms cope with a large service environment. This is achieved by expanding sequence workflow tasks up to 40 and candidate services per task up to 20.

In order to cater for RTT measurements between subset of nodes, we make use of meridian RTT dataset [23] which consist of a set of asymmetric RTT measurements between 1740 peer-to-peer nodes.

After the experiment is set up we compare our algorithms against state of the art approaches based on Genetic Algorithm (SGA) and Particle swarm Algorithm (SPSO). The results are presented in the following sub-sections.

5.1 Fitness

We run all algorithms over 200 generations and observe the fitness value during each generation. Figure 9 shows that both VPSO and NGA outperform SGA and SPSO in finding solutions with better fitness, with NGA demonstrating the ability to find the best solutions and SGA finding the worst solutions. In terms of convergence, VPSO and SPSO converge much earlier than NGA and SGA, This hampered their ability to find solutions as good as NGA.

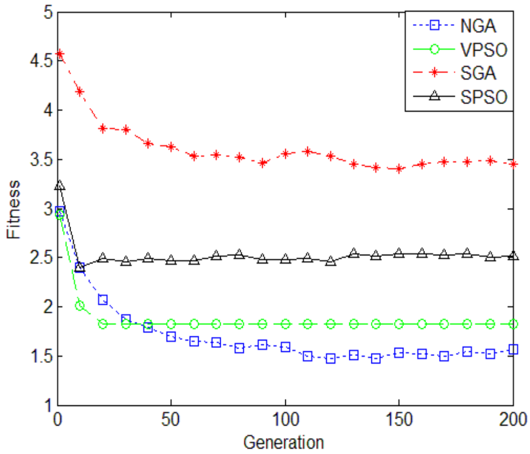

Figure 9: Fitness versus Generations

5.2 Network latency

Here, we compare the network latency of composite services for our algorithms. From Figure 10 it is discovered that VPSO is the best in finding low latency solutions followed by SPSO and NGA, leaving SGA with the worst set of solutions among the lot. VPSO's ability is attributed to uniquely identify and combine population with best network latency (N-population) with other populations. On the other hand, NGA is capable of searching for comparably low latency solutions thanks to its unique mutation operator which utilizes k-means clustering to find low latency solutions without compromising population diversity. The trend observed from SGA, which is a representation of how current evolutionary techniques behave with respect to network latency, shows that without network-awareness in service composition process optimal solutions may have good fitness for cost, response time and reputation but suffer from high network latency which can affect their performance.

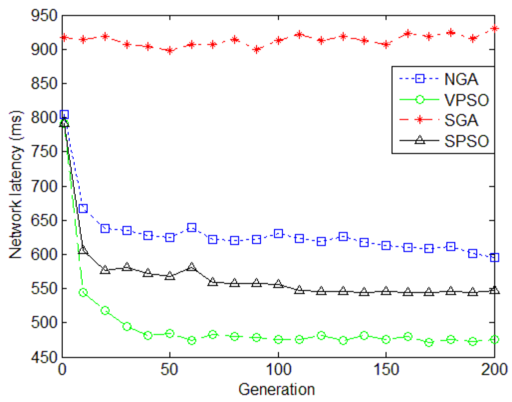

Figure 10: Network latency versus Generations

5.3 Standard deviation

In this experiment, we compare population diversities for the algorithms. Typically the better the population diversity as indicated by standard deviation, the less likely an algorithm will trap into local optimum. Figure 11 indicates that, as expected, VPSO and SPSO demonstrate the poorest diversity hence the reason they converge much earlier than the other two algorithms. NGA shows relatively better diversity while SGA shows the best diversity. This result also indicates that NGA's improved diversity when compared to VPSO is consequence of its novel mutation operator.

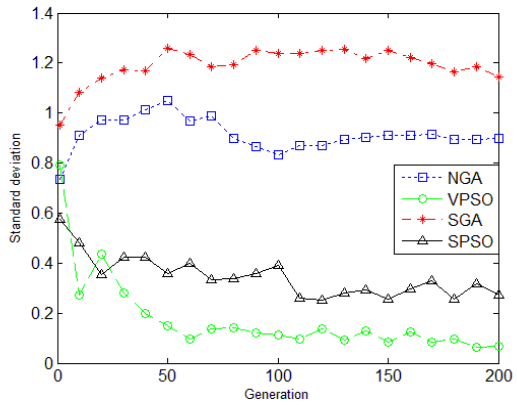

Figure 11: Standard Deviation versus Generations

5.4 Computation time

Table 1 shows that VPSO takes one third of NGA and SGA's computation times to achieve slightly similar results in terms of fitness and network latency. While it takes about half of SPSO's computation time to find significantly better solutions than SPSO. Therefore VPSO is the most efficient amongst the four algorithms, although at the cost of its population diversity. Table 2 indicates that VPSO has the worst fitness of the four algorithms despite obtaining the best network latency while SPSO has the best fitness with SGA showing the worst latency.

Table 1: Computation times (in seconds) of the four algorithms

SGA.	VPSO	NGA	SPSO
110.96s	**30.055s**	109.24s	66.54s

Table 2: Comparison of Algorithms' best results

ALGORITHM	BEST FITNESS	BEST NETWORK LATENCY	BEST STANDARD DEVIATION
NGA	**0.1793**	557.68ms	**1.2949**
VPSO	1.5585	**470ms**	0.9163
SGA	1.0310	574.22ms	1.2895
SPSO	0.6015	531.54ms	0.8326

6 Conclusion

In this paper we propose two evolutionary algorithms that perform network-aware IoT service composition. The aim of the algorithms is to search for composite services with optimum cost, response time, reputation and network latency QoS. The first algorithm is an Evolutionary Particle swarm Algorithm known as VPSO. The algorithm employs evolutionary techniques such as non-domination sort and multi-populations in its operation. The second approach is an N-Genetic Algorithm or NGA. NGA uses a k-means clustering algorithm to classify IoT services into clusters based on their RTT distance to other services and then tries to mutate individuals with other individuals in same cluster. From the results of experimentation, we observe that NGA outperforms in terms of quality of fitness, while VPSO outperforms in terms of computation speed. While both algorithms find low latency solutions, they fall short of population diversity when compared with state of the art Particle swarm and Genetic algorithms, this slight compromise is necessary in order to improve both fitness and network latency. Our results also demonstrate that VPSO is most efficient approach when compared to NGA.

REFERENCES

[1] Martinez-Julia, P.; Torroglosa Garcia, E.; Ortiz Murillo, J.; Skarmeta, AF., "Evaluating Video Streaming in Network Architectures for the Internet of Things," *Innovative Mobile and Internet Services in Ubiquitous Computing (IMIS), 2013 Seventh International Conference on* , vol., no., pp.411,415, 3-5 July 2013

[2] Guinard, D.; Trifa, V.; Karnouskos, S.; Spiess, P.; Savio, D., "Interacting with the SOA-Based Internet of Things: Discovery, Query, Selection, and On-Demand Provisioning of Web Services," *Services Computing, IEEE Transactions on* , vol.3, no.3, pp.223,235, July-Sept. 2010

[3] Jaeger, M.C.; Rojec-Goldmann, G.; Muhl, G., "QoS aggregation for Web service composition using workflow patterns," *Enterprise Distributed Object Computing Conference, 2004. EDOC 2004. Proceedings. Eighth IEEE International* , vol., no., pp.149,159, 20-24 Sept. 2004

[4] Yongjun Liao; Wei Du; Geurts, P.; Leduc, G., "DMFSGD: A Decentralized Matrix Factorization Algorithm for Network Distance Prediction," *Networking, IEEE/ACM Transactions on* , vol.21, no.5, pp.1511,1524, Oct. 2013

[5] Natallie Kokash; "An Introduction to Heuristic Algorithms," *Department of Informatics and Telecommunications,* vol., no., pp.1-8, 2006.

[6] Miranda, V.; Fonseca, N., "EPSO-evolutionary particle swarm optimization, a new algorithm with applications in power systems," *Transmission and Distribution Conference and Exhibition 2002: Asia Pacific. IEEE/PES* , vol.2, no., pp.745,750 vol.2, 6-10 Oct. 2002

[7] Master, Chen Peng; Professor, Xu Guiqiong, "A brief study on clustering methods: Based on the k-means algorithm," *E -Business and E -Government (ICEE), 2011 International Conference on* , vol., no., pp.1,5, 6-8 May 2011

[8] Fenye Bao; Ing-Ray Chen, "Trust management for the internet of things and its application to service composition," *World of Wireless, Mobile and Multimedia Networks (WoWMoM), 2012 IEEE International Symposium on a* , vol., no., pp.1,6, 25-28 June 2012

[9] Spiess, P.; Karnouskos, S.; Guinard, D.; Savio, D.; Baecker, O.; Souza, L.; Trifa, V., "SOA-Based Integration of the Internet of Things in Enterprise Services," *Web Services, 2009. ICWS 2009. IEEE International Conference on* , vol., no., pp.968,975, 6-10 July 2009

[10] Li L; Xinrui, L.; Xinyu, L.;"Cloud-Based Service Composition Architecture for Internet of Things," *Communications in Computer and Information Science,* Springer, vol.312,no.,pp.559-564, 2012.

[11] Chen Ming; Wang Zhen wu;,"An Approach for Web Service Compositon Based on QoS and Discrete Particle Swarm Optimization," Software Engineering, Artificial Intelligence, Networking, and Parallel/Distributed Computing, SNPD 2007, Eight ACIS International Conference on, vol.2, no., pp. 37-41, August 2007.

[12] Edgardo Avilés-López, J Antonio García-Macías;"Mashing up the Internet of Things: a framework for Smart Environments," *EURASIP Journal on Wireless Communications and Networking*, vol.2012, no.1, pp.1, 2012.

[13] Lifeng, Ai; "QoS-aware Web Service Composition Using Genetic Algorithms," *PhD Thesis Queensland University of Technology*, USA on, vol., no., pp.30, 2011.

[14] Qian, L.; Runliang, D.; Fuzan, C.; Guofang, N.; "A QoS-oriented Web Service Composition Approach based on multi-population genetic algorithm for Internet of Things," I*n International Journal of Computational Intelligence Systems*, vol.7, no.2, pp.26-34, 2014.

[15] Farhan, H.; "QoS Based Dynamic Web Service Composition," In *International Journal of Computer Science and Information (IJCSIS),* vol., no., pp., 2010.

[16] Yijie, S.; Gongzhang, S.; "Improved NSGA-II Multi-objective Genetic Algorithm Based on Hybridization-encouraged Mechanism," In *Chinese Journal of Aeronautics,* vol., no.21, pp.540-549, 2008.

[17] Rony Kay; "Pragmatic Network Latency Engineering Fundamental Facts and Analysis," cPacket Networks on vol., no., pp.1-13, 2009.

[18] G. Canfora, M. D. Penta, R. Esposito, and M. L. Villani. An approach for QoS-aware service composition based on genetic algorithms. In GECCO '05: Proceedings of the 2005

conference on Genetic and evolutionary computation, pages 1069–1075, New York, NY, USA, 2005. ACM

[19] Kashif, D.; Amirhosein, T.; Romain, R.; Frank, E.; “Adaptable Service Composition For Very-large-scale Internet of Things Systems,” *in Proceedings of the 8th Middleware Doctoral Symposium*, vol., no., pp.1-6, 2011.

[20] Lixing Li; Zhi Jin; Ge Li; Liwei Zheng; Qiang Wei, "Modeling and Analyzing the Reliability and Cost of Service Composition in the IoT: A Probabilistic Approach," *Web Services (ICWS), 2012 IEEE 19th International Conference on* , vol., no., pp.584,591, 24-29 June 2012.

[21] Liu, J; Yuxi, C; Xu, C; Jianli, D;, “A Cooperative Evolution for QoS-driven IoT Service Composition,” *Automatika Journal for Control, Measurement, Electronics, Computing and Communications*, vol.55, no.4, pp.438-447, 2014.

[22] Adrian, K.; Fuyuki I.; Shinichi Honiden,"Towards network-aware service composition in the cloud," In *Proceedings of the 21st international conference on World Wide Web* (WWW '12). ACM, New York, NY, USA, on, vol., no., pp.959-968, 2012.

[23] Wong, B.; Slivkins, A.; Sirer, E.; “Meridian: A lightweight network location service without virtual coordinates,” *In: Proc. the ACM SIGCOMM.*, vol., no., pp., 2005).

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Quality Assessment of Web Services using Soft Computing Techniques

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ABSTRACT

With the ever growing number of web services in the Internet the selection of a suitable web service has become a mind boggling task. Several quality parameters are being considered for a judicious selection of a web service in a given service context. One of the popular measures for classifying web services is the use of Web Service Relevance Function (WSRF). A number of classifiers have been used which have yielded accuracy up to 99% considering WSRF as one of the attributes of QWS dataset. But, the maximum accuracy achieved by any classifier without WSRF is only 89.99%. In this paper, a feed-forward backpropagation neural network (BPNN) with adaptive momentum factor has been employed to further improve the accuracy. Furthermore, a similar neural network is modelled for determining the WSRF of the web services included in the QWS dataset. As an alternative, Genetic Algorithm is used to find the weight factors associated with each parameter to calculate WSRF, assuming it to be a linear function of those parameters. The average errors for WSRF per pattern obtained in the two approaches are 1.0447 and 1.176 respectively. These calculated WSRF values can be used for classification to enhance the accuracy of a classifier.

Keywords: Web services, Quality of Services (QoS), Backpropagation, Continuous Genetic Algorithm

1 Introduction

Web services are self-contained, self-describing, modular applications that can be published, located, and invoked across the Web [1]. A Web Service is a means of communication between two electronic devices over a network. It is a software function provided at a network address over the web and represents a software system designed to support interoperable machine-to-machine interaction over a network. Different software systems often need to exchange data with each other, and a web service is a means of communication that allows two software systems to exchange data over the internet. The software system that requests data is called a service requester, whereas the one that processes the request and provide the service is called a service provider. An increasing amount of companies and organizations only implement their core business and outsource other application services over the Internet. So the ability to efficiently and effectively select and integrate inter-organizational, heterogeneous services on the Web at runtime is an important step towards development of Web service applications.

The web services, a novel paradigm in software technology, have innovative mechanism for rendering services over diversified environment. They promise to allow businesses to adapt rapidly to changes in the business environment and the needs of different customers. With the increasing use of web services, standardization of basic content integration, support of complex service-oriented architectures, provision of seamless integration of business processes and applications etc. has led to an increase in the number of both web service consumers and providers [2]. The rapid

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introduction of new web services into a dynamic business environment can adversely affect the service quality and user satisfaction. Consequently, assessment of the quality of web services is of paramount importance in selecting a web service for an application. From a service consumer's perspective, Quality of Service (QoS) plays a crucial role while selecting a particular web service from among many alternatives found in the UDDI registry.

2 Related Work

In paper [3], Mohanty et al. developed various classification models based on intelligent techniques namely BPNN, PNN, GMDH, TreeNet, CART, SVM and J48 to predict the quality of a web service based on a number of QoS attributes. They observed that the accuracy with WSRF for GMDH and J48 is maximum i.e. 100%, where the accuracies without WSRF for GMDH and J48 are 89.75% and 67.77% respectively. In case of PNN accuracy without WSRF is maximum i.e. 89.99%. Therefore, there is significant difference in accuracies of classifiers with WSRF and without WSRF. Also, the classification accuracy greatly depends on the WSRF value which helps GMDH and J48 to approach accuracies up to 100%. However, the accuracies of all those techniques fall down miserably while not considering the WSRF as one of the input attributes.

LI Yuan-jie et al.[4] compared 3 classification algorithms, Naïve Bayes, SVM, REPTree, in classifying the WSDL data, and then analyzed the ensemble learning classification trained by AdaBoost. In this paper, they applied automatic web service semantic annotation and used these three classification methods and furthermore ensemble learning is applied. According to the experimental results using 951 WSDL files and 19 categories, the accuracy was 87.39%.

In [5], authors employed Naïve Bayes, Markov blanket and Tabu search techniques to classify web services. They noted that the average accuracy of Naïve Bayes classifier is 85.62%, followed by Tabu search of 82.45% and Markov blanket of 81.36%. In this context, they employed Back propagation trained neural network to find the importance of different attributes in web services and they found that WSRF plays a vital role for classifying the web services. Excluding the WSRF from dataset they observed that the average accuracy of Naïve Bayes is 75.01%, Markov Blanket is 65.48% and Tabu search is 71.38%.

Web service selection has been extensively studied in [6-11]. Xu et al (2007) [12] have presented a QoS-based web service selection approach that is based on ratings indicating the level of client's satisfaction with a web service following certain interaction with it. A drawback of the approach is the assumption that the ratings are objective and valid. Thus, the web service selection process becomes untrustworthy. Al-Masri and Mahmoud (2007) [13] introduce the notion of web service relevancy function in order to measure the relevancy ranking of a particular web service. The function calculates the distance between a particular QoS value and the maximum normalized value in its corresponding set.

In [14], the QoS is utilized to differentiate a single web service, which best meets client requirements among multiple web services with the same functional properties. The QoS data collected during web service executions is used for prediction of the future values of QoS properties, based on probability evaluation. Thus, the client will be able to select a web service with the largest probability of having QoS properties that are as close as possible to the user defined requirements.

The suitability of ANN for ranking quality of web services has been reported in [15, 16]. In [15] performance of various Artificial Neural Network (ANN) training algorithms in predicting the ranking of a web service has been reported. In [16], the average performance rate of ANN recorded is 95%. It is also observed that by removing some of the input parameters of ANN, the accuracy degrades significantly.

3 Methodology

The ease of implementation and greater generalization capability made ANNs popular among the researchers for various applications. In this paper, ANNs has been implemented to address the problem of accurate determination of WSRF parameter of web services. The back propagation algorithm, which is a gradient decent algorithm, is used for training the network in a supervised manner. The Back Propagation algorithm is widely used in many applications. However, this algorithm is sometimes trapped by local minima which limit its efficiency. To overcome this limitation many variations to the standard algorithm has been proposed. These variations to standard backpropagation include introduction of learning coefficient to speed up the learning. The value of the learning rate should be sufficiently large to allow a fast learning process but small enough to guarantee its effectiveness. To avoid oscillation, momentum factor is introduced with small values of learning coefficient. The performance of the algorithm can further be improved by adaptively changing the learning rate and momentum coefficient. In the first phase of this work, backpropagation algorithm is used for quality assessment of web services. Different neural network structures have been modelled for classification of web services and determination of the web services relevance function. As an alternative approach to compute WSRF value, Genetic Algorithm is used for weight determination for each parameter. The QWS dataset is chosen to formulate the objective function. The total absolute error is minimized by optimizing the weight factors of the web service parameters. A simple Genetic Algorithm is described as follows.

Genetic Algorithm (GA) is a global search method based on natural selection procedure consisting of genetic operators such as selection, crossover and mutation. GA optimizers are particularly effective in a high-dimension, multi-modal function, in which the number of variables tend to be higher, for their easy searching process. GA performs its searching process via population-to-population (instead of point-to-point) search. Parallel architecture of GA makes it robust which uses probabilistic and deterministic rules. A member in a population called a chromosome, is represented by a binary string comprising 0, 1 bits. Bits of the chromosome are randomly selected and the length of bit strings is defined in relevance. However, real values are taken in continuous genetic algorithm. In order to apply the methodology, an initial randomly generated population is required. From initial population, child population is born guided by three operators such as reproduction, crossover and mutation. New born child members are judged by their fitness function values. These child members act as parents in the next iteration. This procedure is repeated till the termination criteria are met.

The pseudo code of a genetic algorithm is as follows:

```
Simple Genetic Algorithm ( )

{
        Initialize the Population;
        Calculate Fitness function;
        While (Fitness Value! = Optimal Value)
{
                Selection;
                Crossover;
                Mutation;
                Calculate fitness Function;
        }
}
```

4 Experimental Set Up

The QWS (Quality of Web Service) dataset is chosen for this purpose. Artificial Neural Network is used for two different purposes like (i) classification of web services and (ii) finding the value of WSRF. Continuous Genetic Algorithm (CGA) is used for finding the values of WSRF.

4.1 Data set 1

The QWS dataset consists of data from over 5000 web services out of which the public dataset consists of a random 364 web services which have been chosen. The services were collected using Web Service Crawler Engine (WSCE). The majority of Web services were obtained from public sources on the Web including Universal Description, Discovery, and Integration (UDDI) registries, search engines, and service portals. The public dataset consists of 364 Web services each with a set of nine Quality of Web Service (QWS) attributes that have been measured using commercial benchmark tools. Each service was tested over a ten-minute period for three consecutive days. WSRF is used to measure the quality ranking of a Web service based on the parameters 1 through 9 listed in table-1.

Table 1: QWS Parameters with their description and units

P-ID	Parameter Name	Description	Units
1	Response Time (RT)	Time taken to send a request and receive a response	ms
2	Availability (AV)	Number of successful invocations/total invocations	%
3	Throughput (TP)	Total Number of invocations for a given period of time	Invokes per second
4	Success ability (SA)	Number of responses / number of request messages	%
5	Reliability (REL)	Ratio of the number of error messages to total messages	%
6	Compliance (CP)	The extent to which a WSDL document follows WSDL specification	%
7	Best Practices (BP)	The extent to which a Web service follows WS-I Basic Profile	%
8	Latency (LT)	Time taken for the server to process a given request	ms
9	Documentation (DOC)	Measure of documentation (i.e. description tags) in WSDL	%
10	WSRF	Web Service Relevancy Function: a rank for Web Service Quality	%
11	Service Classification	Levels representing service offering qualities (1 through 4)	Classifier
12	Service Name	Name of the Web service	None
13	WSDL Address	Location of the Web Service Definition Language (WSDL) file on the Web	None

4.2 Data set 2

This is a new version of QWS Dataset that includes a set of 2,507 Web services and their QWS measurements that were carried out using the Web Service Broker (WSB) framework [17]. Each row in this dataset consists of 11 parameters separated by commas for each Web service. The first nine parameters are QWS parameters measured using Web service benchmark tools over a six-day period. The QWS values represent averages of the measurements collected during that period. The last two parameters represent the service name and reference to the WSDL document. Example:

67.5,86,6,86,73,78,80,1.5,95,check,http://ws.cdyne.com/spellchecker/check.asmx?wsdl.

4.3 Platform

Both the backpropagation neural networks and Continuous Genetic Algorithm are implemented in Pentium i5, 1.80 GHz processor, and 4 GB RAM.

5 Results and Discussion

5.1 Classification of web services using Neural Networks

Neural networks are being efficiently used for classifying web services. Since WSRF is the most important parameter of web services, two different networks have been considered for this purpose.

In the first case, P-IDs (Parameter-IDs) 1 to 10 from table-1 have been used as the inputs.

A 10x10x1 network is modelled as shown in figure 1. A single output represents the class of the web service.

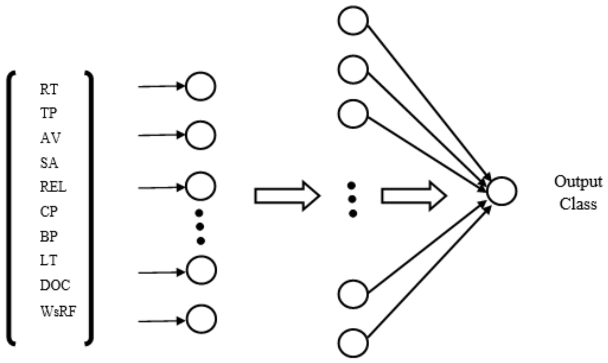

Figure 1: Network Structure

The network is trained for 2000 iterations without momentum coefficient and with variations to momentum coefficient.

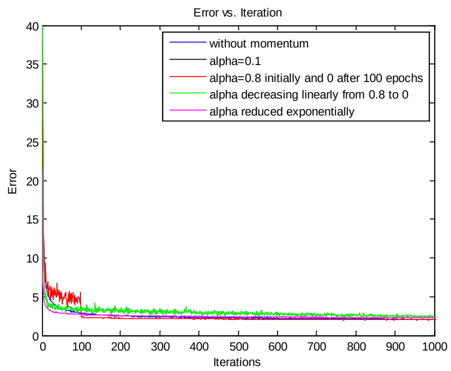

Figure 2: Effect of adaptive momentum coefficient in training NN for classification

The learning constant is taken as 0.04. At first the network is trained without momentum. Then, it is trained with a momentum of 0.1 keeping learning constant 0.04. To introduce adaptive momentum, the momentum variations are considered in three different ways. In the first case, the momentum is kept high at 0.8 for 100 iterations and then reduced to zero. In the second case, the momentum is reduced linearly from high value to zero. Finally, the network is trained with exponentially decreasing momentum coefficient keeping the learning constant at 0.04. From above it is clearly noticeable that use of adaptive momentum improves learning the neural network. It is seen that the learning with exponentially decaying momentum is more effective. A 10 fold validation technique has been used to train and test the network until maximum classification accuracy is attained. In our case we could achieve 100% accuracy. In the second case, a network structure 9x7x1 is modelled for classification of the web services without WSRF parameter from the inputs in figure 1. A similar adaptive momentum coefficient is considered for training the network. Existing classifiers show maximum 89.99% accuracy without considering WSRF as one of the input parameters. The proposed model with 10-fold validation achieves an accuracy of 97.22%.

5.2 Computation of WSRF using Neural Networks

The WSRF is the most important parameter for classifying various web services which depends on other parameters like reliability, Successability, Documentation and Response time etc. However, to our knowledge there is no tool to compute WSRF parameter directly. Even though there exist many detailed guidelines for measuring the quality of web services, there is still a debate about what actually constitute a good web service. Furthermore, there is a lack of empirical validation for good quality web service guidelines. Thus, in our view defining a quality of service metrics is necessary in order to overcome this problem. In this work we have considered the parameters Response time, Throughput, Availability, Successability, Reliability, Compliance, Best Practices, Latency and Documentation to compute the WSRF value using the QWS dataset.

Therefore, a 9x10x1 network is modelled which takes inputs such as Response Time (RT), Throughput (TP), Availability (AV), Successability (SA), Reliability (REL), Compliance (CP), Best Practices (BP), Latency (LT), and Documentation (DOC). The output of the ANN is the WSRF value, which is used to measure the quality ranking of a Web service based on the quality metrics.

The learning coefficient is kept constant at 0.04 and the momentum coefficient is changed adaptively. In order to analyze the effect of adaptive momentum coefficient, the momentum is decremented with training iterations in three ways such as: i) momentum coefficient is decremented linearly from a high value 0.8 to zero, ii) momentum coefficient is kept constant at a high value 0.8 for 100 iterations and zero for the rest, and iii) momentum coefficient is decremented exponentially with iterations.

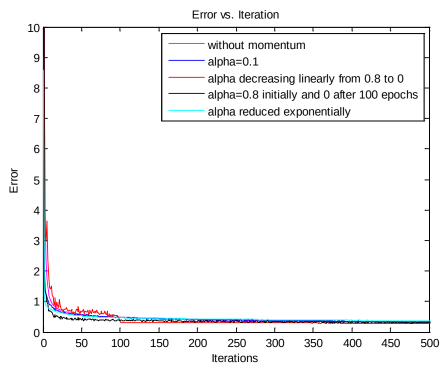

Figure 3: Effect of adaptive momentum coefficient in training NN for WSRF calculation

A sample of calculated WSRF by using neural networks with actual WSRF values is recorded in Table 2. The average error per pattern in testing mode is 1.0447.

Table 2: WSRF as predicted by ANN

Sl. No.	P1	P2	P3	P4	P5	P6	P7	P8	P9	Actual WSRF	Computed WSRF
1	999.71	99	0.8	94	38.5	78	74	837.41	93	67	66.51937
2	2987.9	56	0.2	17	38.8	78	82	2987.9	8	39	41.1091
3	125.44	100	13.5	86	86.4	78	80	125.33	91	86	86.77515
4	326.56	84	6.2	77	15.9	89	89	303.92	5	55	56.56019
5	582.36	28	1.1	16	22.1	89	69	181.86	88	47	46.49747
6	1075.33	100	1.2	75	50	78	84	1014	92	67	68.23855
7	115	86	6.6	57	76.6	78	84	112.5	37	69	69.81351
8	502	24	3.4	9	21.8	78	85	499.28	12	35	36.37946
9	7170.79	85	0.6	89	32	78	77	7170.43	89	62	60.94153
10	2049.5	63	0.4	38	50.7	89	91	2049.4	66	55	55.53001

5.3 Genetic Algorithm for finding WSRF of web services

The QWS dataset of 364 web services is used for this purpose. Since, the parameters response time (p1), availability (p2), throughput (p3), successability(p4), reliability(p5), compliance(p6), best practices(p7), latency(p8) and documentation(p9) vary in units, they are first normalized. It is assumed that the normalized WSRF is a linear function of remaining normalized QoS parameters. Therefore, we can formulate a function for normalized (WSRF)n considering all normalized web service parameters(pi)n values as:

$$WSRF_n = \sum_{i=1}^{n=9} w_i (p_i)_n \tag{1}$$

A continuous genetic algorithm is used to determine the coefficients or weights (w_i). A population size of 80 is chosen. The probability of crossover and mutation are 0.85 and 0.001 respectively. Since some of the QoS parameters contribute negatively to WSRF, the range of nine coefficients is taken [-1 1]. The objective function is formulated by taking 90% of the QWS data, i.e., 328 web services at random. Total absolute error for those 328 records is minimized. The fitness function taken is:

$$Fitness = \frac{1}{(1+E)} \tag{2}$$

Where, E is the total absolute error of 328 web services.

The optimized weights obtained by genetic algorithm are tabulated below.

w1	w2	w3	w4	w5	w6	w7	w8	w9
0.17591 1	0.162317	0.263999	0.115322	0.285133	0.073854	-0.08905	-0.19998	0.20415 9

It is interesting to note that the negative signs with 7th and 8th weights corresponding to best practices and latency indicate their negative effect on WSRF measures. These optimized weights are used to compute WSRF for the remaining 36 web services. Finally, all the values are denormalized to obtain the actual WSRF values. The average error for per pattern is found to be 1.176.

A sample of calculated WSRF using the weights optimized by CGA (Continuous Genetic Algorithm) with actual WSRF values is recorded in Table 3.

Table 3: WSRF as Computed by CGA

Sl. No.	P1	P2	P3	P4	P5	P6	P7	P8	P9	Actual WSRF	Computed WSRF
1	166.11	100	10.3	86	75.7	78	80	158.89	90	81	81.13802
2	308	100	4.2	82	90.3	78	86	305.69	8	66	66.89934
3	105.4	100	16.5	80	89.4	78	72	105.4	91	87	88.984
4	213	83	20.4	50	77.2	78	91	210	42	72	73.40467
5	795.42	88	2.6	76	19.3	78	77	353.17	87	61	61.79505
6	7170.79	85	0.6	89	32	78	77	7170.43	89	62	63.79179
7	615	100	7.8	88	48.5	100	84	599.33	58	71	71.81647
8	141.66	26	9.2	10	87.6	78	86	135.33	12	53	53.96712
9	213.25	78	5.8	67	56.5	89	84	138.25	94	73	71.23432
10	456.75	71	4.3	62	30.3	89	84	432.5	4	51	49.79714

The above result shows that an empirical formula can be established by incorporating the weights, normalization and denormalization of each attribute, as an alternative means to compute the WSRF value as follows:

$$WSRF=70*[5.72329E\text{-}06(RT-45)+ 0.001887407\ (AV-14)+ 0.008979558\ (TP-0.1)+ 0.0012535\ (SA-7)+ 0.003106024\ (REL-5.9)+ 0.002238\ (CP-67)- 0.002406757\ (BP-58)- 6.50352E\text{-}06(LT-31.5)+ 0.002126656\ (DOC-1)]+30 \quad (3)$$

5.4 Comparison of WSRF Calculation by ANN and CGA

In previous two sections, ANN and CGA were used for calculating WSRF with dataset 1. The results are in good agreement with actual values. Here, both the techniques are applied to dataset 2, with similar functionalities as dataset 1 for which WSRF value is not available in order to compare the two techniques. A few samples of the result are tabulated below.

Table 4: Comparison of WSRF Calculated by ANN and CGA

RT	AV	TP	SA	REL	CP	BP	LT	DOC	WSRF(ANN)	WSRF(CGA)	Difference
319.25	89	2.6	96	73	100	80	99.5	10	68.158455	66.76079	1.39767
198	96	14.7	99	67	78	72	50	10	71.086480	72.12520	1.03872
523.75	94	6.9	95	73	100	80	64.5	94	82.666750	82.63907	0.02768
320.48	86	1.2	86	53	89	66	125.18	10	58.665986	60.88265	2.21667
196.33	87	9.5	95	53	100	71	38.12	86	76.558963	79.20620	2.64724
474.91	91	5.8	97	60	78	74	162.87	86	73.921122	75.20932	1.28820
414	86	10.5	86	73	89	84	74.5	4	69.571611	67.21170	2.35991
514.33	47	3	47	73	78	75	123.5	9	52.836561	54.47793	1.64137
174.11	60	7.5	61	60	89	74	104.28	88	70.359578	70.95054	0.59096
205.33	90	3.5	97	60	100	74	48.33	28	69.180367	68.38793	0.79244
2836.25	79	2.4	79	73	67	84	78.75	7	56.793737	58.54982	1.75609

It is seen that the WSRF calculated by the two techniques are very close to each other. Besides accuracy, the proposed empirical formula inferred by CGA is very simple and efficient. Hence, it is much better than a trained neural network to compute WSRF values.

6 Conclusion

In this work, two different back propagation neural network models have been used for classification of web services with constant learning coefficient and adaptive momentum coefficient. In the first model using WSRF as one of the input parameter we achieved a classification accuracy of 100%. The second model which does not consider WSRF as an input parameter could yield only 89.9% accuracy as far as reported by others. However, in our proposed model the classification accuracy has been significantly improved to 97.22%.

Further, ANN is also used to predict WSRF values. The average error per pattern as predicted by the proposed neural network model is found to be 1.0447. As an alternate approach, an empirical formula for calculating the WSRF value is established by using CGA. The average error per pattern in this case is found to be 1.176. Finally, we compared the WSRF values obtained by applying the neural network model and the proposed formula using the dataset 2 with similar characteristics.

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REFERENCES

[1] Jinghai Rao and Xiaomeng Su., “A Survey of Automated Web Service Composition Methods”, Lecture Notes in Computer Science, 2005, Volume 3387/2005, 43-54.

[2] S.Susila and S.Vadivel, “Web service Selection through QoS agent Web service”, IJSWS 13-325; © 2013, pp.18-23.

[3] Ramakanta Mohanty, V. Ravi and M.R. Patra, “Web-services classification using intelligent techniques”, Elsevier, Expert Systems with Applications 37 (2010), pp. 5484–5490.

[4] LI Yuan-jie and CAO Jian, “Web Service Classification Based on Automatic Semantic Annotation and Ensemble Learning”, 2012, IEEE 26th International Parallel and Distributed Processing Symposium Workshops & PhD Forum, 2274-2279.

[5] Ramakanta Mohanty, V. Ravi and M. R. Patra, “Classification of Web Services Using Bayesian Network”, Journal of Software Engineering and Applications, 2012, 5, 291-296.

[6] L. Ai and M. Tang, “A hybrid genetic algorithm for the optimal constrained web service selection problem in web service composition”, IEEE Congress on Evolutionary Computation, Barcelona, Spain, 2010, 8 pages.

[7] E. Maximilien and M. Singh, “A framework and ontology for dynamic Web services selection” IEEE Internet Computing, vol. 8, no. 5, pp. 84–93, Sept.-Oct. 2004.

[8] K. Verma, R. Akkiraju, R. Goodwin, P. Doshi, and J. Lee, “On accommodating inter service dependencies in web process flow composition,” in Proc. AAAI Spring Symposium on SWS, 2004, pp. 37–43.

[9] G. Canfora, M. Di Penta, R. Esposito, and M. L. Villani, “An approach for QoS-aware service composition based on genetic algorithms”, In Proc. of the 2005 conference on Genetic and evolutionary computation., New York, NY, USA: ACM, 2005, pp. 1069–1075.

[10] D. Ardagna and B. Pernici, “Adaptive service composition in flexible processes”, IEEE Transactions on Software Engineering, vol. 33, no. 6, pp. 369–384, June 2007.

[11] Yu, T., Zhang, Y., and Lin, K.J., 2007, “Efficient algorithms for Web services selection with end-to-end QoS constraints”, ACM Trans. Web 1, 1, Article 6 (May 2007), 26 pages.

[12] Xu, Z.; Martin, P.; Powley, W. and Zulkernine, F.; “Reputation-Enhanced QoS-based Web Services Discovery”, 2007, Proceedings of IEEE International Conference on Web Services. Salt Lake City, USA, pp. 249-256.

[13] Al-Masri, E. and Mahmoud, Q., 2007, “Discovering the Best Web Service”, Proceeding of the 16th international conference on World Wide Web. Banff, Canada, pp. 1257-1258.

[14] Petrova-Antonova, Dessislava., "A qos-aware approach for web service selection based on probability evaluation", M CC SSI MCCSIS (2011): 43.

[15] M. Kamalahhasan, “Applications of Neural Networks for Ranking of Web Services using QoS Metrics”, SSRG International Journal of Electronics and Communication Engineering (SSRG-IJECE) – volume 1, Issue 1–Feb 2014, pp. 4-8.

[16] Eyhab Al-Masri and Qusay H. Mahmoud, “Discovering the Best Web Service: A Neural Network-based Solution”, Proceedings of the 2009 IEEE International Conference on Systems, Man, and Cybernetics San Antonio, TX, USA - October 2009, 4250-4255.

[17] Al-Masri, E., and Mahmoud, Q. H., "QoS-based Discovery and Ranking of Web Services", IEEE, 2007, pp.529-534

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Mechanism for reliable low latency communications in Computing Clusters

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ABSTRACT

Recent increases in the demands for computing power have given rise to the prevalence of distributed computing. Computationally complex problems are broken down into smaller chunks and are distributed to computing nodes that perform the computation simultaneously. The nodes may exchange information as peers and their combined result is the final outcome of the computation. Multiple computers need a mechanism to communicate and exchange information in order to harness collective computing power. Traditionally, the Transmission Control Protocol (TCP) has been used to exchange information between computers. However, additional overhead, generally associated with TCP has been considered as a serious drawback for any rapid data exchange. The User Datagram Protocol (UDP) alleviates the overheads of TCP but provides no reliability for the data transfer. In this paper, we introduce an enhanced UDP mechanism to exchange data of limited size reliably and without any associated overhead of TCP.

Keywords: Reliable communications; Cluster computing; Transmission Control Protocol; User Datagram Protocol; Reliable User Datagram Protocol; Overhead in communication protocols.

1 Introduction

TCP protocol was designed to enable reliable communication and optimize the bandwidth between a sender and a receiver. It is a highly robust and versatile protocol, being able to account for varying availability of the network bandwidth between the computing nodes. It is able to adjust the transmission speed on the fly and allows for retransmissions thereby minimizing the data loss and ensuring reliability. TCP achieves this by setting up a communication channel between two communicating nodes, allocating buffers at the sender and the receiver and using sequence numbers to keep track of the packets already transmitted. Reliability is ensured through acknowledgement messages sent by receiver back to the sender [1] [2]. Traditional internet applications are built on the client-server paradigm and thus TCP is well suited for such applications. However, since TCP requires that a dedicated channel is setup between a server and a recipient it cannot provide any support for broadcasting messages. In a multiprocessor environment, broadcast is often required to notify all the participating nodes within a limited time. Setting up a dedicated channel for each node would be both time and resource intensive. The UDP protocol tries to solve these problems inherent in TCP. It does not require the setup of a dedicated channel between the sender and the intended recipient and has support for broadcast [3]. However, UDP does not provide support for acknowledgement messages and therefore the sender has no way of interpreting if the message was reliably transmitted to the recipient. Reliability of transmitted messages is important in large distributed systems since the outcome of a particular computational

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task is the result of all the nodes working in tandem. Failures in a message transmission can stall the computation and erroneous data transmission may even result in unexpected outcomes. In this work, we propose a novel mechanism for fast transmission of data in a multiprocessor environment. Our mechanism ensures the reliability of the data being transmitted between nodes and allows for retransmissions in case of any failure. This mechanism has been tested on the 32 node computational cluster at the University of Cincinnati and could form the core of the networking architecture for fine grain parallelism in large scale distributed simulations.

2 Background

Our extensive experiments with network traffic in computing clusters have indicated that it is inherently highly intermittent in nature. Computing nodes only exchange traffic when they need to communicate with other nodes and/or fetch data from the master node. These exchanges are infrequent and only involve modest amount of data transfers. However, for the purposes of a successful computational task, we need to ensure that this data is transferred reliably with a bounded latency since the outcome of the computation depends on it. We normally use TCP to guarantee delivery of packets in a cluster. Our results illustrate that packet losses increase in direct proportion to the number of packets being exchanged at any given time in the cluster. Since packet drop causes TCP to retransmit, this exponentially increases the number of packets and the performance of the network degrades rapidly.

We used a simple setup in the cluster to simulate the behavior of a distributed computing system with fine grained parallelism. Such a system shares data between nodes on the level of object states across nodes and therefore has substantial number of data packets being exchanged when running. We simulated a lightly and a heavily loaded system on the cluster to examine the effects to traditional TCP under heavy intermittent traffic. For the lightly loaded system simulation, the nodes generated a packet every half second. The heavily loaded system simulation had nodes generating ten packets every second. For distributed systems involving hundreds of nodes, the packet transmission rates can approach the rates in our simulation. We exchanged 25,000 packets to simulate a sustained data transfer. The effect of using TCP for a lightly loaded and a heavily loaded system are shown in Figures 1 and 2 respectively. We used a timeout of 3 seconds for the lightly loaded system and 8 seconds for the heavily loaded one. We see from the figures that the heavily loaded system has substantially more packet losses than the lightly loaded one even with the higher timeout value. While most packets were received and acknowledged, the heavily loaded system shows a far higher proportion of lost packets. The white areas in the graph represent packets that were not received at the receiver required retransmission. From the graphs, we can infer that TCP is unable to deal with the requirements of intermittent network traffic even with a small payload. More precisely, our results indicate that since TCP is heavily geared towards stream based traffic, the overhead for setting up a TCP connection between a sender and a receiver far often exceeds the actual data that needs to be transferred over the established TCP channel. On the other hand while UDP allows fast data transmission and support for broadcasting, it does not provide support for acknowledgements and thus lacks in providing reliability. In this work, we have implemented the best of both transmission protocols by adding reliability on top of the UDP protocol.

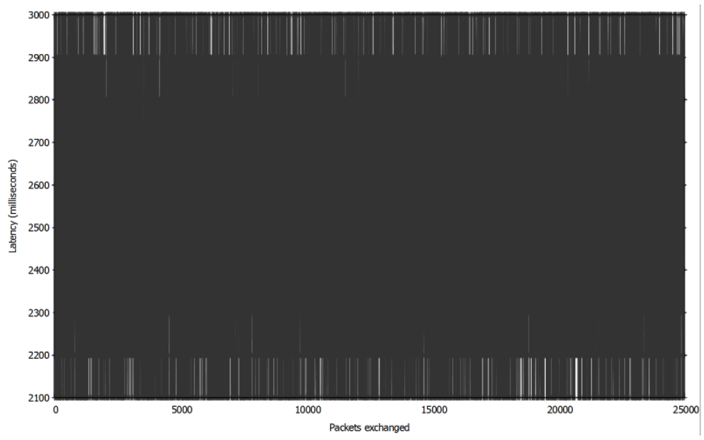

Figure 1: Latency and associated packet drops for our lightly loaded cluster.

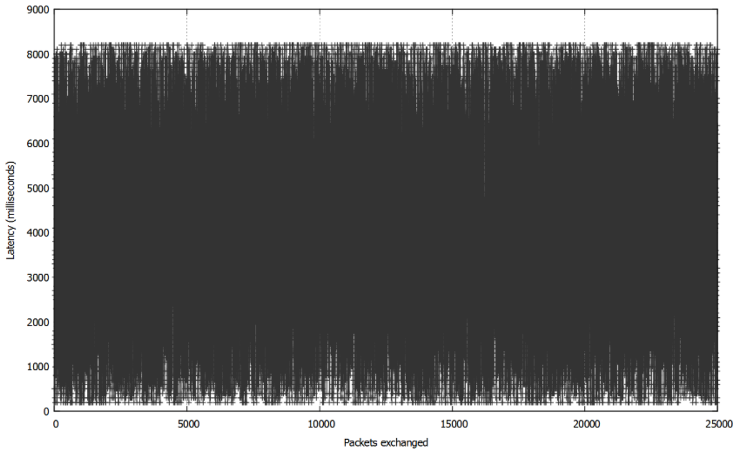

Figure 2: Latency and associated packet drops for our heavily loaded cluster.

3 Related Work

Studies have been performed in the past to improve and alleviate the limitations of the UDP protocol. Extensive analysis of the UDP protocol implementation has been performed in the UNIX kernel [4].Through protocol optimization, UDP has been shown to perform over 25-35% better in CISC and RISC systems [5]. Extensions have also been made to the protocol to support wireless transmissions and its usage in wireless multimedia sensor networks since wireless networks are more susceptible to packet losses due to the nature of the wireless medium [6] [7]. Efforts have also been made to improve the reliability of UDP. The most notable of these is the Reliable User Datagram Protocol (RUDP) designed at Bell Laboratories [8] [9]. RUDP attempts to achieve a middle ground between the simplicity of UDP and the overheads incurred by TCP. However, in order for RUDP to achieve a higher quality of service, it builds on the UDP by adding support for acknowledgement of received packets, mechanism for retransmissions of lost packets, flow control and adding support for buffers at the communicating endpoints [9]. Cisco uses an implementation of RUDP in its Signaling Link Terminals. RUDP Version 0 is used to backhaul the Signaling System 7 protocols over IP signaling control networks while Version 1 is used for ISDN networks [10] [11]. The low latency provided by UDP makes it suitable for IPTV and VoIP applications since they require very high throughputs and are somewhat tolerant to low levels of packet losses [12]. With the prevalence of VoIP applications in recent years, UDP currently forms a significant portion of the total internet traffic [13]. RUDP also forms the basis of the communications architecture in the MediaRoom IPTV platform originally developed by Microsoft which is now owned by Ericsson [14]. However, both Cisco and Microsoft/Ericsson implementations of RUDP are proprietary and are used in their own products, making them incompatible with existing network infrastructure. To the best of our knowledge, this is an implementation of a general purpose low latency mechanism for fast transmission of packets with limited payload.

4 Our mechanism

4.1 Scope

Our mechanism as described in this work is specifically geared towards communication between nodes in a computing cluster. In order to make our mechanism compatible with existing networks, we have built it on the top of existing network paradigms. Specifically, it covers the Transport, Session and the Presentation layers of the communication stack. Computing nodes use IP addresses to identify and talk to each other. Our mechanism also uses the IP addressing scheme that is fully compatible with existing network architectures. We use UDP as the core of the protocol as it provides a better throughput as compared to TCP as we have shown earlier. Reinventing and implementing a completely new and separate transport layer protocol would have put in too much overhead and made our mechanism incompatible with current standards. Moreover, using UDP provides our mechanism with broadcasting support that is very useful in a cluster network. We have built on UDP's fast transmission capabilities and have implemented a mechanism that informs the sender when a packet has been received by an intended recipient. We use sender side buffer to store the information sent over the network until an acknowledgement is received from the recipient. The sender uses identification numbers on the data packets to achieve this. We understand that while the majority of packets that are expected to be exchanged using our mechanism would be small enough to fit inside a UDP datagram, certain scenarios may give rise to heavy traffic in a cluster. Therefore, we have incorporated functionality in our mechanism that is able to split a large payload into fragments that can be made to fit inside a conventional UDP datagram. This functionality ensures that our mechanism works under all traffic scenarios in a computing cluster.

4.2 Architecture

Our mechanism follows conventional network paradigms and defines a customized header for each message. The actual data is carried in the message payload and is fully described in the message header. The 25 byte header in our mechanism completely specifies the parameters for the sender and the receiver including the IP addresses and the port numbers. A schematic diagram of the message header is shown in Figure 3. As mentioned previously, we use sequence numbers to keep track of the messages sent. In order to keep our mechanism simple we choose the initial value of the sequence number as a pseudorandom number and keep incrementing the value with every packet that is sent. However, any monotonic function that varies with time can be used for this purpose. We identify each message uniquely by a combination of the sender and receiver and the message id.

Message Size (16 bits)	Message Sequence No. (16 bits)	Multipart message Field No. (32 bits)	Number of Messages (32 bits)	Message Id (32 bits)	Message Data Size (64 bits)	Bit Fields (8 bits)

Figure 3: Structure of the Header.

The header fields determine all the functionalities of our mechanism. However, the header overhead is minimal as we show in the results. We describe the usage of each field in the header in Table 1.

Table 1: Header fields and their usage

Field	Usage
Message Size	Stores the size of the complete message including the message header.
Message Sequence Number	Determines the order of messages originating from a particular sender. A pseudorandom number is used as the seed with a monotonic function for successive sequences. See Section 4.2.
Multipart Message Field Number	Determines the position of a particular message in a multipart message. Since messages can arrive out of order in a packet switched network, the receiver thread uses the information in this field to rearrange the message parts in the receiver buffer before handing over to the higher layers. Obfuscates message transmission details from higher network layers and enables compatibility with existing network protocols.
Number of Messages	Used only for multipart messages and is used to specify the total number of messages that are in a particular message. The receiver thread uses the information in this field to determine when all the parts of the multipart message have arrived successfully at the receiver. The Recombination thread starts once all the parts of the multipart message have been successfully received. See Section 4.3.4.
Message Id	Uniquely identifies a particular message in the list of all messages exchanged in the cluster.
Message Data Size	Stores the size of the payload in a particular message.
Bit Fields	Used to specify added functionality at the receiver. A combination of these fields can be used to signify the start or end of a session. Can also be used to specify no message confirmations based on a particular simulation scenario. See Section 4.3. Broadcast messages and cumulative acknowledgements for network traffic reduction can also be signified using these fields.

4.3 Implementation

We have implemented this mechanism as a multithreaded process running on the computing nodes. Since the nodes need to send and receive messages, the process is capable of running as a client and a server simultaneously. We would now give a detailed overview of the buffers that we mentioned earlier. We would then explain the process flow with respect to the workings of the buffers. All of these buffers are maintained by the process running on each node.

4.3.1 Session buffer

The session buffer is responsible for storing information related to all other nodes that a particular node needs to communicate with. It stores the value of the next sequence number and the timeout interval which is used to determine if a packet has been lost and need to be resend. If the sender does not receive an acknowledgement within the specified timeout, we assume that the packet has been lost and needs to be resent by the sender. We adaptively change the value of the timeout interval based on the current network conditions. We follow an exponential back off strategy for messages that need to be resent. This follows from conventional network protocols that have used this strategy to determine the network load and thereby prevent message collisions. For each new message that is sent or received, a new entry is made in the session buffer. This is to prevent ambiguous messages being received during a network storm. We also mark an entry with a broadcast flag if the message has been received as part of a broadcast.

4.3.2 Transmission buffer

The transmission buffer is responsible for arranging the messages that are to be sent. The memory allocated in the buffer is used to mark the headers and fill the payload section with the actual data that is to be transferred. The messages are removed from the transmission buffer once acknowledgements have been received from the receiver.

4.3.3 Reception buffer

The reception buffer is responsible for receiving the messages and forming an ordered list. This is essential in order to guarantee that the packets are received in the order in which they were sent. Due to the nature of the packet switched network, messages might arrive out of order at the receiver. A separate thread running on the receiver looks at these messages and arranges them in the order based on the sequence number of the messages. Once the messages have been arranged in the specified order, they are made available to be used by the higher layers of the network stack.

4.3.4 Multipart message buffer

The multipart message buffer is used for handle large payloads and to ensure compatibility with the existing network infrastructure. As mentioned previously, when the sender has a large message to send, it is split into a number of parts before transmitting to the receiver. The partial message buffer allows the messages to be gathered at the receiver while they are being sent and rearrange them by means of the multipart message field number as mentioned previously. Once the whole message has been reassembled at the receiver, the destination application is notified of the arrival. Providing this functionality frees the application layer from having to rearrange parts of a multipart message in order. This mechanism also provides compatibility with older application layer protocols that do not have functionality to assemble multipart messages

4.4 System Walkthrough

We have a custom Linux application written in C++ that spawns multiple threads on the computing node to implement the mechanism that we have described in this work. In this section we describe three major threads that are responsible for transferring the data from the application layer, sending acknowledgements and performing memory allocation and management of the buffers. The server thread forms the bridge between the processes running on the node and our mechanism. When an application is ready to send data, it notifies the server thread which then proceeds to add it to the transmission buffer. This thread also maintains the timeout counter. Once the counter expires and the server thread finds no acknowledgement for a particular message that had been sent earlier, it resends the message again to the intended recipient. The transmission buffer is cleared by the server thread once an acknowledgement is received from the recipient node.

Once a recipient node receives a message, it first verifies if the message is already present in the reception buffer. This situation can arise if the acknowledgement was lost and the originator node had resent the message. This functionality is performed by the receiver thread with the help of the message id. If the message is not present, the message is added to the reception buffer. However, if the receiver thread detects that the message is already present, it is an indication that the sender of the message has not yet received the confirmation. It then proceeds to resend the confirmation message to the sender. For multipart messages, the receiver thread has an added responsibility of storing the received messages in the multipart message buffer. As mentioned previously, messages in a multipart message may arrive out of order. It is also the responsibility of the receiver thread to arrange all the parts of the multipart message. Once the receiver thread receives all the parts of the multipart message, it sends a confirmation to the sender. A cumulative confirmation for a multipart message suffices to inform the sender that all parts of the message have been received.

A buffer manager thread continuously runs in the background and is responsible for maintaining the buffers at an optimum load level. Too many packets in the buffer could mean that either the application layer is generating too many packets or there is some congestion in the network. Based on the values of the timeout counter, the buffer manager can determine the cause of the backlog. If the network is healthy, the buffer manager spawns new server and receiver threads to send and process the messages that are being generated. If it determines congestion in the network, the server manager thread informs the application that it needs to slow down or pause the generation of messages since most of the messages have to be retransmitted anyways. Informing the application is necessary since in a computing cluster, nodes work on data that is generated in a different node and thus they need to be synced periodically. If the node pauses or slows down computation, then the synchronization procedures can be performed without a large rollback penalty.

We use a publish/subscribe model to notify the application thread of the arrival of messages and the current health of the network. Each application can subscribe to be notified of arriving messages that match a predefined criterion. The receiver thread maintains a table of all possible matches for a particular node. Once a message satisfying a particular criterion arrives at the receiver, the particular application is notified of the arrival. Similarly, applications also subscribe to be notified of network events. Network congestion and severe packets loss scenarios are propagated to the application layer from our mechanism. We also provide error handlers for messages who have no subscribers and for common error codes. The publish/subscribe model allows us to decouple our mechanism from the application layer. We leave it to the particular application to decide on the course of action based on a particular error code or the prevailing network scenario.

A schematic diagram showing the flow of control for the mechanism described in this section is shown in Figure 4.

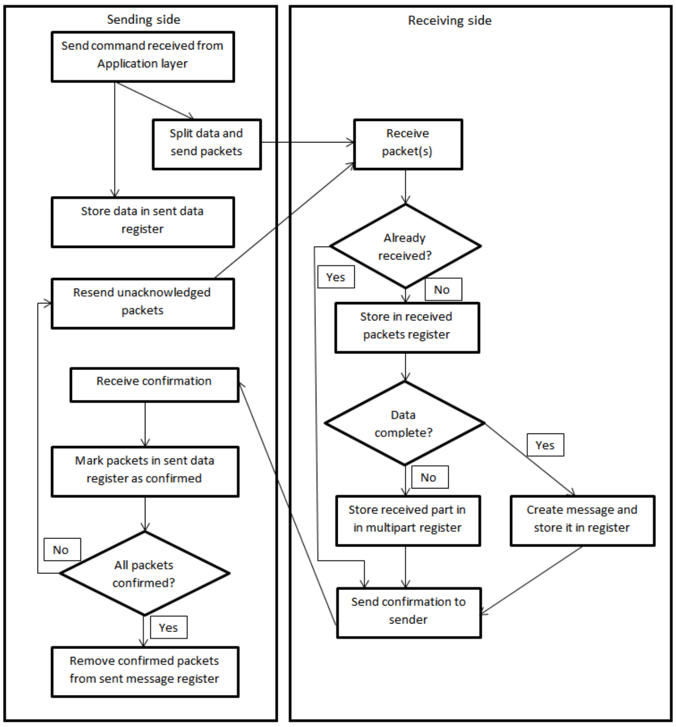

Figure 4: Control flow for our mechanism.

5 Results

We used the mechanism as described in this work at the University of Cincinnati computing cluster with 32 nodes. The nodes are synchronized from the same network time server. We exchanged messages ranging from a zero payload (only the header) to 5 Megabytes. Large message exchanges are also carried out in order to ensure that we could test the implementation of the multipart messaging scheme as described in Section 4.3. The nodes are programmed to exchange a million messages for the purposes of this study. In order to compare our mechanism we also measured the latencies using TCP and UDP as the communication protocol. The messages contained timestamps from the server and were used by the receiver to compare the latencies.

We understand that the network traffic encountered at any given time in the cluster would be a function of the number of nodes in the cluster, the current state of the computation and the number of signaling messages being exchanged at a particular instant. Generally, nodes exchange a lot of data during the start of a particular computation task thereby giving rise to high latencies. As the computation proceeds, the number of messages exchanged reduces since each node tends to work on their own set of data using their own memory. Towards the end of the computation, the number of messages exchanged increases since the nodes need to synchronize the results between themselves, weed out erroneous results and transmit all the computed work back to the master node. We used an average of the million latency measurements for each payload data size in order to eliminate any stray errors. We also simulated node failures and rollbacks with random bursts of heavy network traffic to temporarily increase the instantaneous latency and test how our mechanism holds out in such scenarios. Our results for 2, 3, 4 and 5 nodes are shown in Figures 5, 6, 7 and 8 respectively.

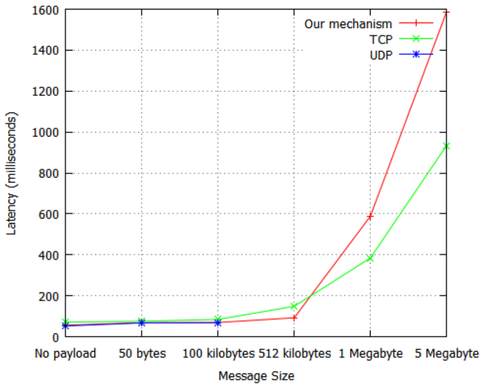

Figure 5: Latencies for messages exchanged between two nodes

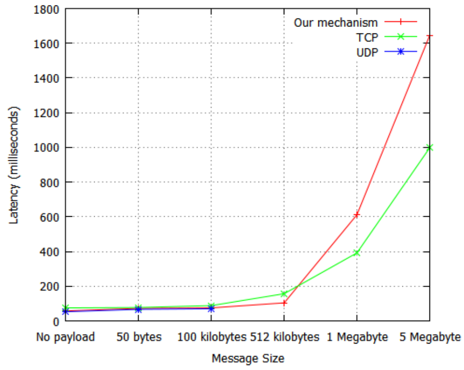

Figure 6: Latencies for messages exchanged between three nodes

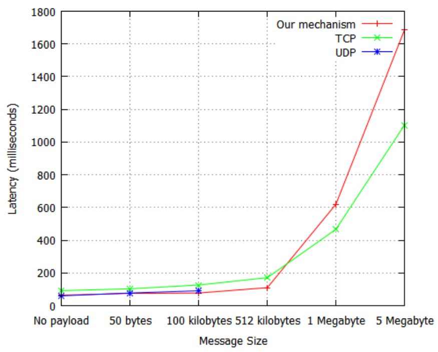

Figure 7: Latencies for messages exchanged between four nodes

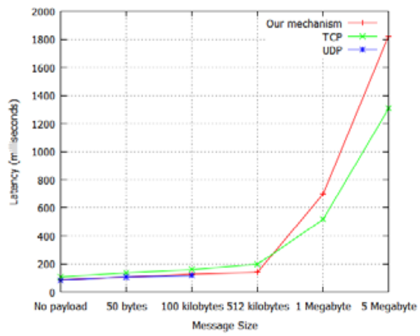

Figure 8: Latencies for messages exchanged between five nodes

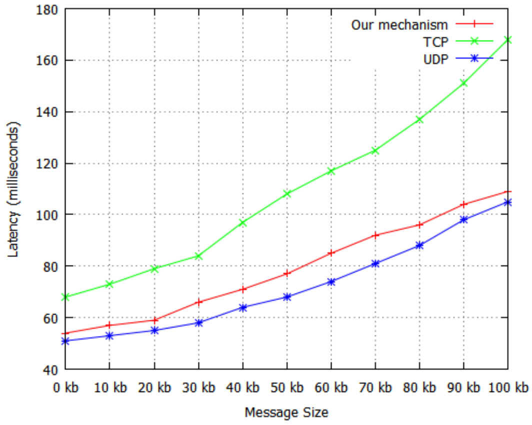

Figure 9: Latencies for small message payloads between two nodes

6 Conclusion

We found that for small payloads, the latencies for our mechanism matched very well with those of the UDP. However as discussed in this work, the overhead for our mechanism is almost negligible. Also for small payloads, our mechanism has almost half the latency of TCP. As we have mentioned earlier, almost all message exchanges in a computing cluster have an extremely small payload. Therefore, our mechanism takes the best of both the TCP and the UDP for most scenarios.

The results for our mechanism indicate that the latency for data transmissions carried out using our mechanism does not increase appreciably as the number of nodes in the cluster is increased. However, the latency for TCP increases exponentially. This is because, as the number of nodes increases, the number of channels that TCP needs to establish also increases, thereby degrading the performance. However, since our mechanism does not need to setup channels, its overhead does not depend on the number of nodes in the network. Hence the performance would not drop drastically unlike TCP. Therefore, latencies experienced by our mechanism are not dependent on the number of nodes and thus could hold good for large clusters having hundreds or thousands of nodes.

For very small payloads, we observe that the graphs for our mechanism and those of TCP and UDP are almost overlapping. This is because of the large differences in the data sizes used in the experiment. For the purposes of clarity, Figure 9 is a detailed summary of the observed latencies for small payloads.

We also observe from the graph that TCP performs exceptionally well for large payloads. However, we can also see that our mechanism does not fail for large payloads even with the added complexity of sending, receiving and combining multipart messages and providing for reliability. Therefore, we can conclude that our mechanism can be used as a replacement for the generic networking mechanism in computational clusters involving a significant number of nodes.

REFERENCES

[1] Cerf, V., and R. Kahn, "A Protocol for Packet Network Intercommunication," IEEE Transactions on Communications, Vol. COM-22, No. 5, pp 637--648, May 1974.

[2] Postel, J., “DOD Standard transmission control protocol,” ACM SIGCOMM Computer Communication Review, Vol. 10, No. 4, pp 52-132, Oct 1980.

[3] Postel, J., User Datagram Protocol, RFC Editor, 1980

[4] Balasubramanian, S., Thanthry, N., Bhagavathula, R. and Pendse, R., "Performance analysis of UNIX user datagram protocol implementations," System Theory, 2004. Proceedings of the Thirty-Sixth Southeastern Symposium on , vol., no., pp.230,235, 2004

[5] Partridge, C. and Pink, S., "A faster UDP [user datagram protocol]," Networking, IEEE/ACM Transactions on , vol.1, no.4, pp.429,440, Aug 1993

[6] Wang, Y., Wu, C., Zhang, X., Li, B and, Zhang, Y., "An embedded wireless transmission system based on the extended user datagram protocol (EUDP)," Future Computer and Communication (ICFCC), 2010 2nd International Conference on , vol.3, no., pp.V3-690,V3-693, 21-24 May 2010

[7] Aghdam, S.M., Khansari, M., Rabiee, H.R. and Salehi, M., "UDDP: A User Datagram Dispatcher Protocol for Wireless Multimedia Sensor Networks," Consumer Communications and Networking Conference (CCNC), 2012 IEEE , vol., no., pp.765,770, 14-17 Jan. 2012

[8] RFC 908, The Reliable Data Protocol (RDP).

[9] RFC 1151, The Reliable Data Protocol (RDP) Version 2.

[10] Modarressi, A.R. and Skoog, R.A., "An overview of Signaling System No.7," Proceedings of the IEEE , vol.80, no.4, pp.590,606, Apr 1992

[11] Cisco Signaling Link Terminal http://www.cisco.com/en/US/products/hw/vcallcon/ps2152/products_data_sheet09186a0080091b58.html

[12] Saleh, S., Shah, Z. and Baig, A., "Capacity analysis of combined IPTV and VoIP over IEEE 802.11n," Local Computer Networks (LCN), 2013 IEEE 38th Conference on , vol., no., pp.785,792, 21-24 Oct. 2013

[13] Zhang, M., Dusi, M., John, W. and Chen, C., "Analysis of UDP Traffic Usage on Internet Backbone Links," Applications and the Internet, 2009. SAINT '09. Ninth Annual International Symposium on , vol., no., pp.280,281, 20-24 July 2009

[14] The MediaRoom IPTV Platform http://www.ericsson.com/ourportfolio/mediaroom-landingpage

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Revisiting the Possible Creation of the Quantum Information Unit-A Necessary Element of Quantum Computation Procedure

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ABCTRACT

In the present work, a previously reported idea has been further developed. The idea is based on the earlier proposed visualization of Shor's states. Therefore, the paper deals with some ideas of quantum physics, which constitute the basis of a set of the quantum objects, employed as q–bits, their possible structure and composition being disclosed. Quantum limit is implemented in cases where the characteristic of the quantum length defined by the de Broglie wavelength is equal to or lower than the physical size of the object. It is this condition should somehow be reflected in the composition and structure of the kantovyh Islands, which should serve as qubits. In other words, in the technological sense of dimensional quantum effects in this particular case, it is necessary to under-stand both the need for such facilities in the process of materializing States Shore, which would match the size (dimensions) of individual quantum islet and a critical length, describing it (the island) structure and (or) property. Also, necessary operations, performance of which at the technological level will allow for electronic unit to organize quantum computation procedure, are analyzed. It is shown that if operations with matrixes are used, the process of quantum computation can take reasonable amount of time.

Keywords: Quantum computer; Shor's cells; matrix algebra.

1 Introduction

Analysis of the state-of-the-art of quantum computations, and in particular, the problems re-lated to creation of quantum computer, shows that the current researches are mainly focused on the search for new ideas, sometimes, absolutely unexpected is put in the forefront. For this reason, our research is mainly of exploratory and, hence debatable character. They relate, first of all, to search for such molecular (multielectron) systems, the application of which should ensure the creation of really operating quantum computer [1]. Besides, one should bear in mind that over the last forty – fifty years, a huge number of new compounds with diverse physical and chemical properties has been syn-thesized and studied.

All (or almost all) researches, which have been carried out by the moment, are based on the search for appropriate specific quantum objects. It is assumed that the development of computation quantum schemes using these objects is rather routine work. In other words, the researchers try to find a solution of the quantum computer problem by studying the behavior of a small number of quan-tum objects. Figuratively speaking, the researchers try to find approaches to the solution of quantum information problem, using "from down to up" ideology. In this direction, a huge amount of work has been performed. In this connection, we should mention the recently published paper

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[2]. It is also re-ported on the methods for the preparation of materials, which could become, at least in principle, a basis for the creation of quantum computers [3]. An important feature is that the problem of quantum computer creation, if to follow the above-mentioned approach, involves mainly the search for appro-priate quantum system (or quantum systems). Creation of really operating quantum processor, suita-ble for realization of the quantum computation procedure, is a matter of technique, i.e. this is selec-tion of special conditions, for example, placing quantum objects in strong magnetic fields using NMR spectrometers.

But eventually the quantum computer, if created, represents also macroobject operating in macroworld. Therefore, "from up to down" seems also to be possible. This approach assumes that the creation of devices, suitable for quantum computations, is based on the initial application of systems containing many quantum particles and is not reduced to behavior of small ensembles composed of either already existing or specially synthesized materials. Probably, only in this case, we can use ad-vantages of the quantum computer entirely. Otherwise, taking into account great advancements in the field of modern (classical) computers development, there is no sense to invest the efforts into so-lution of the quantum computation problem. Moreover, the latter inference puts forward a problem of compatibility of the quantum computer with classical one, if to bear in mind the scale of classical computers applications and their ample opportunities. The conclusion is as follows. At the present, real quantum computer is the additional unit which is built in modern classical computers. When needed, such unit can significantly accelerate the computation procedure. However, quantum computer can also be employed in parallel computations, when this unit connected to the general system of classical computer performs calculations at some stage of the solution of a specific problem. At the same time, main units of the classical computer can solve next tasks. Apparently, it is pertinent to remember here that in the work [4], the idea (minimum program) about allocation of databank, obtained using a sys-tem of elementary quantum computers, for expansion of possibilities of Turing universal computer, has been formulated.

The paper [5] is also devoted to the above-mentioned problems. The authors analyze the state-of-the-art of experimental researches in the field of quantum computer creation and justify the idea of Shor states visualization as the first step on the way of creation of real quantum processor. The idea is based on application of the phenomenon of secondary wave generated by incident electromagnetic radiation on the object which undergoes screening. According to the theory of wave flow, the incident primary electromagnetic irradiation under certain conditions can be transformed into waves of other type, for example, surface plasmons. The massif from n quantum objects generating secondary irradi-ation, in principle, can be considered as a route to materialization of Shor cells. In other words, a sys-tem of q-bits as a base of real quantum computer can be realized in such a way.

2 Possible Functional Characteristics of Quantum Objects intended for Visualization of Shor's States

The real Shor's states are quantum-physical in essence. Therefore, the massif of quantum is-lets used for their materialization (visualization) should retain this physical feature, i.e. obey to the laws of quantum physics. It is obvious that today among such objects are heterosystems with reduced dimension or low-dimensional structures, in which the motion is limited in one, two or three direc-tions. It is well known that quantum limitation is realized when the characteristic quantum length, de-fined by de Broglie wavelength becomes equal or lower than the corresponding physical size of object. This condition should be somehow realized in structure and composition of quantum islets, which should play the role of q-bits. In other words, from technological point of view, manifestation of size quantum effects in this specific case should be understood as a need for providing of such conditions in the course of a materialization of the Shor's states, when the size (sizes) of individual quantum inlet and some critical length characterizing structure and (or) properties of this islet should coincide. For example, these are size of magnetic domains, length of free path of the charge carrier (let us say, elec-tron), optimum size of crystallites and their structure, etc. Understanding of this fact should lead to active (not passive) participation of quantum islets in the organization of calculation procedure. Correspondingly, the external effect (from macroworld) resulting in incorporation of quantum information unit in general electronic scheme of the computer operation, should have the relevant nature.

As to realization of quantum computation procedure, the process, when each materialized (visualized) Shor's cell corresponds to an element of the electronic scheme, for example, nanotransistor or nanodiode, is supposed to be the most natural. Here it should be emphasized that a trigger scheme has been already proposed. This scheme is based only on graphene ribbons of about 10 nm width, in which all elements are created due to the combination of the specified ribbons tapes, which are cut out under different angles or have different width and/or type of edge [6]. Then such elements are united in the corresponding scheme capable of performing the computation procedure. Both elements of this scheme and the scheme itself should operate according to the laws of quantum physics. Being a real quantum system, such a device performs real computation procedures similar to classical computer, but at the level of microworld.

According to the theory, the Shor's states are resulted from entanglement of q-bit states (including entanglement of quantum dots, if the latter act as q-bits). Therefore, they essentially differ from bits of the classical computer. Consequently, an ij-state can be employed in the calculation procedure over certain period of time after the states i and j will take part in the calculation. This is applicable to all ij-states which should play a role of Shor's cells. To clarify this statement let us suppose that at the given moment, for example, 5j state, i.e. the state of q-bit with i=5, is involved in the calculation procedure (j changes from 1 to m, except j = i). This means that such q-bit already ceased to be entangled with the state of q-bit (j – 1), but it is not yet entangled with the state of q-bit (j + 1). Hence it follows that implementation of the Shor's cells is defined by a strong condition, namely there should be a mechanism realized in microworld and obeyed to the laws of quantum physics, which "surveille" the states of q-bits system and ensures their entanglement when they are involved in the calculation procedure. In turn, existence of such mechanism should be somehow reflected at the macroworld level in order to one can use results of the calculation performed by the system of quantum bits (q-bits). The same problem arises also in the case of states of individual quantum points, i.e. ii-states: their involvement in the calculation procedure should be somehow ordered.

Taking into account the above remarks, the conditions for application of a massif of quantum islets (for example, 10^{10} in number) as q-bits for the organization of calculation procedure can be presented as follow. The initial moment is the assumption that such massif is incorporated into ordinary (classical) computer as separate device. At a certain stage of the computer operation, the electromagnetic radiation generated by any element of the computer excites secondary radiation, for example, plasmonic oscillations of certain frequency in quantum islets. Let there is a delay line providing for excitation of secondary oscillation in i-islet with certain delay relative to (i-1)-islet. Thus, the operation sequence of elements of the electronic device connected with a massif of quantum islets acting as q-bits will be achieved. Excitement of the secondary radiation should be accompanied by input of some initial data in order to the electronic device can perform necessary calculations. Results of the calculation can be entered into the scheme of the classical computer using a device similar to the unit which provides operation of quantum calculation.

It has been reported [5] that via simple scanning of *i* values one can obtain the cells corresponding to own values (a_1, a_2, …) of the ($\hat{\mathbf{A}}$) operator of physical value, which characterizes such processes in the quantum computer as, for example, current passage or wave process. At the same time, mathematical resource of the quantum computer is associated also with application of non-diagonal matrix elements, i.e. a_{ij}. If to follow this logic, this purpose requires, first, the corresponding set (massif) of quantum objects, second, comparison of the specified objects with the corresponding elements of the electronic scheme and, third, provision of consistency or sequence of usage of the quantum objects constituting other massif.

Formation of another massif of quantum objects for the purpose of visualization is not obligatory at all, if to use their combination. According to the laws of quantum physics, if the system consists of n two-level q-bits, it generally represents superposition of 2n basic states. From this fact follow the main advantages of the quantum computer (see, for example, [7]). If to be limited by rather small amount of quantum particles ($n = 10^2$), one can obtain quite large big mathematical information resource of the quantum computer.

$$2^n = 2^{100} \approx 10^{30}$$

This is the number of Shor's states, visualization of which is described in the present paper. The massif from 10^{10} islets and their combination by two allows reaching this purpose (at least, in principle). If to start from the fact that both individual quantum islets and their combinations by two will be involved in the operation, it is needed that secondary radiation would contain another frequency, which should ensure the application of ij-states for realization of calculation procedure. Naturally, in this case, it is necessary that the sequence of such states usage would be provided (in our case, 10^{20}).

Further specification of the possible unit for quantum computation will be connected with selection of chemical composition and molecular structure of the material, from which quantum islets should be prepared. The compounds with unoccupied 3d-, 4f- and 5f-shells are supposed to be appropriate for this purpose. Peculiarities of their electron and spatial structure of such compounds are mainly defined by interelectron correlations. Of special importance are paramagnetic complexes, which are characterized by intramolecular rearrangement, so-called valence tautomerism. Qwing to this rearrangement, molecular states with essentially different magnetic properties can be realized. If, for example, there are two such states, in the from 10^{10} quantum objects, one of these states will be present in every period of time $5 \cdot 10^9$, and approximately the same amount in the second state (at room temperatures). The specified states can be used for the number coding (for example, zero and unit) [1, 8].

All the above-stated concerning visualization of the Shor's states and specific characteristics of the quantum islets used for this purpose (of course, in the most general view) can be considered as physical modeling of these states. This modeling can be used for organization of quantum computation procedure, i.e. the problem can be solved at the technological level. In its turn, it is logically to pass from this technological level to, figuratively speaking, the mathematical description or representation. The two massifs discussed above (individual quantum objects and their combination) can be presented as two matrixes. Therefore, at some stage of the calculation procedure, these matrixes can be subjected to mathematical operations, thus expanding computing opportunities of the quantum unit. Indeed, operation with two matrixes will lead to a third matrix, which can be involved in operations with two first ones, etc. It is known that matrix algebra is well-developed branch of mathematics. Opportunities of the matrixes allow describing diverse physical processes using numerical actions over the elements of these matrixes. This, in turn, will make it possible to obtain a set of specific numerical data, which can be further used. Next, they can be applied for creation of the corresponding matrixes, which can be processed independently upon solutions of specific objectives.

The material discussed in the present paper is a logical framework based on theoretical representations (in particular, of quantum physics). In our opinion, this represents a first and necessary step (together with the ideas stated in the manuscript [5]) which allows to create, at least, an experimental sample of the device which then can help to solve the problem of quantum computation performance on a wide scale.

3 Conclusion

The real Shor's states are quantum-physical in essence. Therefore, the massif of quantum islets used for their visualization should retain this physical feature, i.e. obey to the laws of quantum physics. It is obvious that today among classical representatives of such objects are heterosystems with reduced dimension or low-dimensional structures, in which the motion is limited in one, two or three direc-tions. Quantum limitation is realized when the characteristic quantum length, defined by de Broglie wavelength, becomes equal or lower than the corresponding physical size of object. This condition should be somehow realized in structure and composition of quantum islets, which should play the role of q-bits. From technological point of view, manifestation of size quantum effects should be un-derstood as a need for providing of such conditions in the course of a materialization of the Shor's states, when the size (sizes) of individual quantum inlet and some critical length characterizing struc-ture and (or) properties of this islet should coincide. Understanding of this fact should lead to active participation of quantum islets in the organization of calculation procedure.

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REFERENCES

[1] Voronov, V.K., Quantum computer: problems and possible solutions. 2014, Saarbrucken: Palmarium Academ-ic Publishing, Deutschland. 93.

[2] Zhaokai, Li, et al., Experimental Realization of Compressed Quantum Simulation of 32-spin Ising Chain.Physical Review Letters, 2014. 112: p. 220501- 220505.

[3] Aldoshin, S.M., et al., On the way to the creation of materials for quantum computers. Ushekhi Khim., 2012. 81 (2): p. 91-104.

[4] Voronov V.K., NMR and the Problem of Quantum Computer Creation: New Outlook, in: S.Shannon (ed.), Trends in Quantum Computing Research, NOVA Publishers: New York, 2006, p. 73 – 90.

[5] Voronov, V.K., A Possibility of Application of Wave Flow Method for Visualization of Shor States. Int. J. Communications, Network and System Sciences, 2013. 6: p. 485-488.

[6] Sorokin P.B., and L.A. Chernozatonsky, Semiconductor nanostructures on Graphene osnovae, Uspekhi Phys. Nauk. 2013. 183(2): p. 113-132.

[7] Valiev K.A., Quantum computers and quantum computing. Uspekhi Phys. Nauk. 2005. 175 (1): p. 3-39.

[8] Voronov V.K., Possible schemes of calculation modeling in a quantum computer, Natural Science. 2010. 2(8): p. 923-927.

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Software methods for fast hashing

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ABSTRACT

The carry-less multiplication instruction, PCLMULQDQ, is a relatively recent addition to the x86-64 instructions set. It multiplies two binary polynomials of degree 63, using $GF(2)$ arithmetic, and produces a polynomial of degree 126, stored in a 128-bit register. PCLMULQDQ is intended to speed up computations in $GF(2^{128})$, which are used for AES-GCM authenticated encryption. We show here how PCLMULQDQ can be used for efficient software implementation of a 64-bit hash function that has a low collision probability. While a 64-bit hash is normally not a meaningful security primitive, the discussed hashing algorithm can be leveraged for other usages that enjoy fast hashing, e.g., querying/maintaining databases. On the latest Intel architecture (Codename Broadwell), our hash function can process messages at the rate of ~ 0.13 cycles per byte.

Keywords: hashing, universal hash functions, fast software implementations, PCLMUQDQ.

1 Introduction

Intel architectures introduced instructions that accelerate AES (AES-NI), and supplemented them with the carry-less multiplication instruction $PCLMULQDQ$. It was motivated by the desire to accelerate AES-GCM authenticated encryption, but other usages (e.g., CRC computations) were also considered (see details on $PCLMULQDQ$ and its applications in [5, 6]).

In this paper, we are interested in a non-cryptographic hashing algorithm that has a low collision probability, and can be computed efficiently. In particular, we seek a 64-bit hash. Hash functions with a relatively short digest have various usages, and a leading one is searching/updating databases. For such usages, the hashed messages are relatively short. Typical entries from database columns are, for example, zip code, name, address, salary, age, employer, and inventory. The associated message lengths are: 5 bytes (zip code), 10 bytes (telephone number), 32 bytes (city name).

There are many known (and used) hash functions. Two examples are LOOKUP3 [3], and Google's CityHash [2] (additional examples can be looked up, for example, in [1]). To the best of our knowledge, the collision probability for these hash functions is either verified empirically, and/or depends on some assumptions on the properties of the messages.

In this paper, we discuss a 64-bit hash function that has a provable low collision probability, and can be computed efficiently on modern computer platforms.

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2 Preliminaries and notation

Let n be a positive integer, denote $s = n - 1$, and let $G = GF(2^n)/P(x)$ be the finite field with 2^n elements, represented via the irreducible polynomial $P(x)$ (of degree n). A field element $A \in G$ is a binary polynomial of degree s, $A = A(x) = \sum_{j=0}^{s} a_j\, x^j$ where $a_j \in \{0,1\}$. We also view A as the string of n bits, $a_s a_{s-1} \dots a_0$ (by convention a_s is at the leftmost position). For $A, B \in G$, $A = A(x) = \sum_{j=0}^{s} a_j\, x^j$ and $B = B(x) = \sum_{j=0}^{s} b_j\, x^j$, define addition and multiplication as follows:

1. Addition (+): $C = A + B$ is the polynomial $C = C(x) = \sum_{j=0}^{s} \left((a_j + b_j) \ mod\ 2 \right) x^j$. The string representation of C is the bitwise XOR $A \oplus B$.
2. Multiplication ($\otimes$): $C(x) = A(x) \otimes B(x) = A(x) \times B(x) \ mod\ P(x)$, where "×" denotes polynomial multiplication with coefficients added/multiplied in $GF(2)$. Specifically, if $T(x) = A(x) \times B(x)$, then $T(x)$ is the binary polynomial of degree $2s$, with coefficients t_i satisfying $t_i = \left(\sum_{j=0}^{i} a_j b_{i-j}\right) mod\ 2$ for $0 \le i \le s$ and $t_i = \left(\sum_{j=i-s}^{s} a_j b_{i-j}\right) mod\ 2$ for $s < i \le 2s$.

When $n = 8m$, an element $F = F(x) = \sum_{j=0}^{s} f_j\, x^j \in G$ can be represented in a compact form, by a sequence of m bytes $B_{m-1}B_{m-2} \dots B_0$, and written in hexadecimal notation.

Hereafter, we focus on $n = 64$ and choose the irreducible polynomial $P(x) = x^{64} + x^4 + x^3 + x + 1$. We call a 64-bit string a "quadword".

To illustrate, we offer the following example.

Example 1: Use $n = 64$, $P(x) = x^{64} + x^4 + x^3 + x + 1$.
The element $F = x^{63} + x^4 + x^3 + x + 1$ is written in hexadecimal (compact) notation as the 8 bytes string $800000000000001B$. If $A = FFFFFFFF0000000F$, $B = FFFFFFFF0000010E$, then $A \times B = 55555555555555AA000000FF00000F5A$, and $A \otimes B = 000000FF00000615$.

2.1 The instruction $PCLMULQDQ$

The instruction $PCLMULQDQ$ operates on two 128-bit registers e.g., $xmm1$, $xmm2$ (the registers' names are arbitrary, and the second operand can also be in a specified in a memory location). It executes polynomial multiplication of two polynomials of degree 63, and writes the result (which is polynomial of degree 126) into a third 128-bit register (see precise definitions in [5]). An "immediate" byte selects the two 64-bit halves from $xmm1$ and $xmm2$, which are to be multiplied. For example, $VPCLMULQDQ \ \ xmm0,\ xmm1,\ xmm2,\ \ imm = 0x10$ operates on $xmm1[127{:}64]$ (top half of $xmm1$) and on $xmm2[63{:}0]$ (bottom half of $xmm2$), writing their carry-less product into $xmm0$. With A, B from Example 1 (embedded in two xmm registers), invoking $VPCLMULQDQ\ xmm0,\ xmm1,\ xmm2,\ 0x01$ with the registers contents $xmm1 = 0000000000000000FFFFFFFF0000000F$ and $xmm2 = FFFFFFFF0000010E0000000000000000$, results in the value $55555555555555AA000000FF00000F5A$ in $xmm0$.

3 Efficient hash functions with $PCLMULQDQ$

To define the message space, over which we use $PCLMULQDQ$ and carry out computations in $GF(2^{64})$, we need each message to be a string of quadwords. To this end, we pad messages of arbitrary lengths (in bytes) to the 8 bytes boundary, with zero bytes. In addition, to distinguish messages of variable lengths, we also append a quadword that encodes the message's length. The exact procedure is defined in the following subsections.

3.1 Message padding and length encoding

Consider a message MSG whose length, in bytes, is $lbytes$, and write $lbytes = 8u + v$, where $0 \leq v < 8$. We encode $lbytes$ as a 64-bit quadword, denoted LEN. For example, if $lbytes = 4096$, then $LEN = 0000000000001000$ (in hexadecimal notation).

If $v > 0$, we pad MSG up to the 8 bytes boundary, with $8 - v$ zero bytes. After the (conditional) padding, the padded message consists of $\ell_1 = \lceil lbytes/8 \rceil = u + 1$ quadwords, say $\overline{MSG} = X_1, X_2, \dots, X_{\ell_1}$ (concatenated quadwords, in this order). Finally, we set

$$M = \overline{MSG}, LEN = X_1, X_2, \dots, X_{\ell_1}, LEN$$

M consists of $\ell = \ell_1 + 1$ quadwords. We call M the "formatted message".

Example 2: Take a 9 bytes message $MSG = 090807060504030201$. Here, $lbytes = 9 = 8 \cdot 1 + 1$. This implies that $\ell_1 = \lceil 9/8 \rceil = 2$, and $X_1 = 0908070605040302$, $X_2 = 0100000000000000$. Also, $LEN = 0000000000000009$. Therefore,
$\overline{MSG} = 09080706050403020100000000000000$, and the formatter message is $M = \overline{MSG}, LEN = 0908070605040302, 0100000000000000, 0000000000000009$ (ℓ =3 quadwords).

3.2 The two hash functions

Let ℓ_{max} be a fixed value, and assume that the message space is the set of formatted messages of length $\ell \leq \ell_{max}$ quadwords. Let MSG be a message and let M be its formatted message, consisting of ℓ quadwords. Denote $M = X_1, X_2, \dots, X_\ell$. Consider a key K that consists of ℓ_{max} quadwords $K = K_1, K_2, \dots, K_{\ell_{max}}$ (each K_j is a quadword). We define two hash functions (S, T) for M, using the key K, as follows.

$$S = S(M, K) = \sum_{j=1}^{\ell} X_j \times K_j \qquad (S \in \{0,1\}^{128}) \qquad (1)$$

$$T = T(M, K) = \sum_{j=1}^{\ell} X_j \otimes K_j \qquad (T \in \{0,1\}^{64}) \qquad (2)$$

The function T is the well-known "inner product" XOR-universal hash function, defined over $(GF(2^{64}))^{\ell_{max}}$. Note that $T = S \ mod \ P(x)$.

3.3 Properties of S and T

Proposition 1. Suppose that the $K_{\ell_{max}}$ quadword keys $K_1, K_2, \dots, K_{\ell_{max}}$ are selected independently, uniformly at random, from $\{0,1\}^{64}$. Let M be a (formatted) message of $\ell \leq \ell_{max}$ quadwords (not all of them are zero). Then,

$$Prob \ (S(M, K) = 0^{128}) < Prob \ (T(M, K) = 0^{64}) = \frac{1}{2^{64}} \qquad (3)$$

Remark 1. T is an "inner product" universal hash function (technically, the family of functions $T\ (M, K)$ is universal). Proposition 1 is a well-known property. The use of the quadword LEN, to account for the message length, is essential in order to ensure "suffix-freeness". To illustrate, note that without appending LEN, a message of a single quadword Q would have the same digest as the message of 2 quadwords $0000000000000000, Q$. This does not happen for the formatted messages. Since $T = S \ mod \ P(x)$, it follows that $S = 0 \Rightarrow \ T = 0$, so the number of messages that zero S is smaller than the number of messages that zero T.

Remark 2. Due to the linearity of S (and T), and to Proposition 1, the probability that two different messages M_1, M_2 have the save value $T = TS(M, K)$ is $\frac{1}{2^{64}}$.

Remark 3. Suppose that K_0 is chosen uniformly at random from $\{0,1\}^{64}$, and define $K_j = (K_0)^j \ mod \ P(x), j = 1, \ldots, \ell$. Then, for a nonzero message M of length ℓ quadwords, we have

$$Prob \ (S(M, K) = 0^{64}) \leq \frac{\ell}{2^{64}}$$

This bound is obtained by the following argument. S is an evaluation of a single variable polynomial in the field , whose degree is ℓ (K_0 is it variable). As such, it can have at most ℓ roots in the field.

3.4 The computational cost of computing S and T

Software running on processors that support $PCLMULQDQ$, can compute S by means of ℓ invocations of $PCLMULQDQ$, followed by no more than $\ell - 1$ XOR operations (depending on the method for XOR-ing the intermediate products; note that the $PXOR$ instruction can XOR 128-bit strings in a single 1-cycle invocation). To compute T, it suffices to compute S, and subsequently reduce it modulo $P(x)$.

3.4.1 Estimating the computational cost of computing S

Suppose that the instruction $PCLMULQDQ$ has latency of L cycles, and throughput 1 (the generalization to a different throughput is straightforward). Throughput 1 implies that the processor is capable of dispatching a $PCLMULQDQ$ instruction every cycle (if data to feed the instruction is available). Code that computes S can be written so that its flow interleaves (as much as it can) the operations and assures that data is available to feed $PCLMULQDQ$ every cycle. Consequently, it is theoretically possible to compute ℓ polynomial multiplications ("× ") in $\ell + L - 1$ cycles. Since the processors that we discuss here can execute two $PXOR$ operations in parallel to $PCLMULQDQ$, we neglect the additional overhead of the XOR operations required for the summation. We conclude that the rate at which the ℓ back-to-back "×" multiplications can be executed is $\frac{\ell+L-1}{8\ell}$ cycles per byte (C/B hereafter), which approaches $\frac{1}{8}$ C/B for large ℓ. Therefore, the best throughput for computing S, which we can expect, is 0.125 C/B. To illustrate, note that the latency of $PCLMULQDQ$ is $L = 7$, and the throughput is 1, on the latest Intel architecture Codename Broadwell (BDW hereafter), so $\frac{\ell+L-1}{8\ell} \sim 0.148$ C/B already for $\ell = 32$. The results shown in Section 4, indicate that the theoretical performance can be approached in practice.

3.4.2 Estimating the computational cost of computing S

Deriving T from S requires one reduction step (modulo $P(x)$). An efficient algorithm can carry out the reduction by means of 2 $PCLMULQDQ$ invocations, and it is demonstrated (as a real code snippet) in Listing 1 of the Appendix. On BDW, this reduction algorithm consumes (only) $\sim$ 17 cycles.

A code example for computing S (and T) is provided in Listing 2 of the Appendix.

3.4.3 Using the hash functions

For a real application, it is reasonable to assume that an upper bound on the message length (thus on ℓ_{max}) is known in advance. Therefore, an application that needs to compute (efficiently) many hashes, can generate, as a setup phase, ℓ_{max} keys, and hold them in memory for the computations. This involves generating, uniformly at random, $64 \cdot \ell_{max}$ bits, but the cost is amortized over many hash computations. Often, the computed hash digests are ephemeral, so the keys do not need to be stored (to a disk) and retrieved in subsequent sessions. However, in cases they need to be stored, it is possible to fix one (randomly selected) seed ($seed$) and derive the keys by using some pseudo-random function. One example is to define $K_j = AES_{seed}(j)$. An alternative is to use powers (in the field) of a single key, as in Remark 3, but this comes at the expense a larger bound (depending on ℓ_{max}) for the collision probability.

4 Results

To test the efficiency of our hash functions, we wrote optimized code and measured its performance on the following three latest Intel processors: Architecture Codenames Sandy Bridge (SNB), Haswell (HSW) and Broadwell (BDW).

In particular, we investigated the effect on the observed performance of the hash function, of the different latency and throughput of $PCLMULQDQ$. The relevant $PCLMULQDQ$ latency/throughput values on these processors are: $14/8$ cycles in SNB, $7/2$ cycles in HSW, and $7/1$ cycles in BDW.

The performance results are reported in Table 1. It shows the performance of computing S and T for short (from 1 bytes) messages up to 4KB messages, where the performance is already at its asymptotic value. For large messages, we see that the performance approaches its theoretical limit. For example, the reported code achieves 0.13 C/B on BDW, where the theoretical limit is 0.125 C/B. The small gap between the achieved and the theoretical performance, can be attributed to overheads such as, for example, function calls, data movement/alignment, and pointers arithmetic. As expected, the performance on HSW and on SNB is almost linearly proportional to the PCLMULQDQ throughput.

For long messages, we see that the difference between computing S and T is negligible. However, for short message, the reduction overhead is noticeable, and leads the different costs for computing S and versus computing T.

To test the collision properties, we wired the function to the Google Murmurhash test harness in [4] (that tries to "challenge" hash functions). As expected, we did not find any collisions in a millions of tests.

Table 1. The performance of computing S and T on three different architectures.

$lbytes$		1	5	10	16	20	32	64	128	1024	4096
	Cycles										
SNB	S	70	73	77	59	87	78	115	184	1,179	4,588
	T	46	49	53	39	61	59	97	167	1,161	4,571
HSW	S	50	54	54	18	57	22	29	45	270	1,053
	T	34	38	38	16	41	18	24	38	263	1,044
BDW	S	48	52	49	16	54	20	21	28	140	541
	T	34	38	36	14	40	17	19	27	139	541
	C/B										
SNB	S	69.68	14.58	7.72	3.68	4.34	2.43	1.79	1.44	1.15	1.12
	T	45.52	9.86	5.29	2.42	3.07	1.83	1.51	1.31	1.13	1.12
HSW	S	50.32	10.84	5.42	1.15	2.85	0.70	0.46	0.35	0.26	0.26
	T	34.32	7.66	3.84	1.01	2.07	0.58	0.38	0.30	0.26	0.26
BDW	S	48.32	10.46	4.93	1.01	2.70	0.63	0.33	0.22	0.14	0.13
	T	34.36	7.64	3.58	0.89	2.02	0.54	0.30	0.21	0.14	0.13

5 Discussion

This paper discussed two very simple hash functions that have two important properties: 1) They have a low collision probability; 2) On the modern processors, they have efficient software implementations. These hash functions can be useful in applications that manage databases.

For comparison, we point out that it is possible to compute 32-bit and 64-bit CRC's with $PCLMULQDQ$, and to approach the theoretical performance limit of 0.125 C/B (for large messages). However, the collision rate characteristics of such hashing strategies are completely different from what we have for S and T (and depend on what is assumed on the messages). Indeed, testing verified that a CRC hashing strategy stumbles on collisions for the same sets of messages that were used for challenging S and T (where we found no collisions).

The use of $S \in \{0,1\}^{128}$ (instead of $T \in \{0,1\}^{64}$) is more efficient for very short messages, and this can be considered as the preferable approach in such cases. This approach trades the improved performance with extra storage for longer hash digests. We point out that simply truncating S to 64 bits also gives a 64-bit hash. However, the bound on the collision probability is no longer guaranteed with this approach.

We conclude with two options to achieve further optimization for short message.

1. A single reduction step (see Listing 1 of the Appendix) consumes (on BDW) $\sim$ 17 cycles because the latency of $PCLMULQDQ$ is 7 cycles. However, the effect of $PCLMULQDQ$'s latency can be significantly reduced for cases that require computing hashes of multiple messages, hence

involving multiple reductions. This can be achieved by aggregating multiple hash computations (and reductions), and interleaving the operations in the software flow, to get efficient pipelined execution.

2. Note that our padding method pads a message to the 8 bytes boundary, and then appends an additional quadword that encodes the message length.

    1. The use of the LEN quadword can be avoided completely if an application requires hashing messages with a fixed length.

    2. For variable length short messages, we can “compress” the length encoding. For example, suppose that all the messages are shorter than 256 bytes (which is a reasonable assumption for databases). Then, the lengths of hashed messages can be encoded by a single byte. Now, consider a 5 bytes message. It will be padded with 2 zero bytes and appended with a 1-byte length encoding. Hashing it will require only 1 field multiplication, instead of 2 field multiplication that the current scheme involves.

REFERENCES

[1]. Choosing a Good Hash Function, http://research.neustar.biz/2011/12/05/choosing-a-good-hash-function-part-1/ http://research.neustar.biz/tag/city-hash.

[2]. CityHash, https://code.google.com/p/cityhash.

[3]. R. Jenkins, http://burtleburtle.net/bob/c/frog.c.

[4]. SMHasher MurmurHash, https://code.google.com/p/smhasher.

[5]. S. Gueron and M. E. Kounavis. Intel Carry-Less Multiplication and Its Usage for Computing The GCM Mode, Rev 2.01. Intel Software Network. http://software.intel.com/sites/default/files/article/165685/clmul-wp-rev-2.01-2012-09-21.pdf.

[6]. S. Gueron and M. E. Kounavis. Efficient Implementation of the Galois Counter Mode Using a Carry-less Multiplier and a Fast Reduction Algorithm. Information Processing Letters 110: 549–-553 (2010).

APPENDIX

Listing 1. Software flow for reducing a 128-bit polynomial (string) modulo $P(x) = x^{64} + x^4 + x^3 + x + 1$. (note the assembly AT&T syntax, where the destination register is the rightmost operand) The flow uses two invocations of PCLMULQDQ. $P(x)$ is encoded as `poly = 0x1b`.

```
vpclmulqdq   $0x01, .Lpoly(%rip), ACC, T0     # reduction phase 1
vpand        .Land(%rip), ACC, ACC
vpxor        T0, ACC, ACC
vpclmulqdq   $0x01, .Lpoly(%rip), ACC, T0     # reduction phase  2
vpand        .Land(%rip), ACC, ACC
vpxor        T0, ACC, ACC
```

Listing 2. Software flow (C intrinsics) for computing S.

```
uint64_t UNIVERSAL_HASH_64_C(void *in, unsigned int len, uint64_t const_in){
__m128i ACC, DATA, T0;
__m128i KEY;
__m128i POLY = _mm_set_epi64x(0x00,0x1b);
__m128i ANDMASK = _mm_set_epi64x(0, 0xffffffffffffffff);
int len_save = len;
uint64_t rest = 0;
uint8_t *key_ptr = (uint8_t*)ks;
ACC = _mm_setzero_si128();

while(len>=16)
{
  DATA = _mm_loadu_si128(in);
  KEY = _mm_loadu_si128((__m128i*)key_ptr);
  ACC = _mm_xor_si128(ACC, _mm_clmulepi64_si128(DATA, KEY, 0x00));
  ACC = _mm_xor_si128(ACC, _mm_clmulepi64_si128(DATA, KEY, 0x11));
  in+=16;
  key_ptr+=16;
  len-=16;
}
if(len>=8)
{
  DATA = _mm_cvtsi64_si128(*(uint64_t*)in);
  KEY = _mm_cvtsi64_si128(*(uint64_t*)key_ptr);
  ACC = _mm_xor_si128(ACC, _mm_clmulepi64_si128(DATA, KEY, 0x00));
  in+=8;
  key_ptr+=8;
  len-=8;
}
if(len)
{
  uint8_t *r_ptr = (uint8_t*)&rest;
  while(len--)
  {
    *r_ptr++ = *(uint8_t*)in++;
  }
  DATA = _mm_cvtsi64_si128(rest);
  KEY = _mm_cvtsi64_si128(*(uint64_t*)key_ptr);
  ACC = _mm_xor_si128(ACC, _mm_clmulepi64_si128(DATA, KEY, 0x00));
  key_ptr+=8;
}
DATA = _mm_cvtsi64_si128(len_save<<3);
KEY = _mm_cvtsi64_si128(*(uint64_t*)key_ptr);
ACC = _mm_xor_si128(ACC, _mm_clmulepi64_si128(DATA, KEY, 0x00));
T0 = _mm_clmulepi64_si128(ACC, POLY, 0x01);
ACC = _mm_and_si128(ACC, ANDMASK);
ACC = _mm_xor_si128(ACC, T0);
T0 = _mm_clmulepi64_si128(ACC, POLY, 0x01);
ACC = _mm_and_si128(ACC, ANDMASK);
ACC = _mm_xor_si128(ACC, T0);
return _mm_cvtsi128_si64(ACC) ^ const_in;
}
```

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Routing Protocols under different Mobility Models, Node Density and Speed

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ABSTRACT

Mobile Ad hoc NETwork (MANET) is self-organized and decentralized network. Due to these properties, they are usable in the field of disaster management, war scenarios, vehicular network etc. To send packets from source to destination, route(s) are required; to achieve this task routing protocols are designed. Routing protocols for MANET have been grouped into four types on the basis of single channel classification. These are Proactive, Reactive, Hybrid and Location-based/ Geographic. The performance of these protocols depends on various factors including mobility pattern/model, speed of node, density of node, area, transmission range etc. In this paper the factors taken into account are mobility models, speed of node and density of node. The performance metrics includes throughput, end to end delay and packet loss. The protocols simulated are DSR, LAR, OLSR and ZRP, choosing one from each group discussed above. The mobility models employed in this study are Random Way Point, Gauss Markov, Reference Point Group Mobility and Manhattan Grid.

From the results it is evident that with the change in mobility pattern, speed and density of the nodes the performance varies. Furthermore it has been observed that location based routing protocols (LAR) and OLSR shows good performance with these variations.

Keywords: MANET, DSR, LAR, OLSR, ZRP, Random Waypoint mobility model, Gauss Markov model, Manhattan Grid model, RPGM, End to end delay, Throughput, Packet loss

1 Introduction

Mobile Ad-hoc NETwork (MANET) is an infrastructure less network which can be rapidly deployed with improved flexibility and reduced cost. A node in the network receives data from source node in a multi-hop pattern. The nodes are free to move randomly and may join or leave the network. Due to this element of randomness, the network topology becomes unpredictable and may change rapidly. A routing protocol is implied to find a path from source to destination and many have been proposed in literature so far. These protocols are further divided into four groups depending on their method to find a route.

1. Proactive protocol: The route information is obtained a priori and stored in a table for future lookup. E.g. OLSR [3].
2. Reactive protocol: The route information is calculated when and wherever required hence favoring an on demand route formation. E.g. DSR [1].
3. Hybrid protocol: It combines the advantages of proactive and reactive routing. The route is initially established with some proactive protocol and then to serve the

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demands from additionally activated nodes are responded with reactive flooding. E.g. ZRP [4].

4. Geographical Protocol: The route information is obtained using the location information of nodes. E.g. LAR [2].

There are many challenges with MANET. Due to node mobility links can be made or breached owing to limit in range of communication. The routing protocol should quickly adapt to network changes and find new paths that avoid the failed links. But this is a difficult task due to constraints like low wireless bandwidth and limited battery power of nodes. This overhead can have a significant impact on the overall performance of the network.

The movement of the nodes can be possibly defined by a mobility model. A lot of models have been proposed so far including Random waypoint model [5], Gauss Markov [6], Manhattan Grid [8], Reference point group mobility[7] etc. Each of these models has different ways to define the possible movements of node. The choice of mobility model affects the performance of routing protocols.

These networks are suitable in under water networks and emergency situations like disaster management, war scenarios etc.

A variety of comparative studies have been carried out on routing protocols and mobility models using metrics like packet delivery ratio, end to end delay, packet loss and energy consumption. But there are other factors also, on which the performance depends including traffic pattern, network topology and obstacle positions etc. For the sake of simplicity many studies have used Random way point model, to evaluate the performance of protocols.

In this study, we present a comparative study of Proactive, Reactive, hybrid and geographic protocols. The comparative studies available mostly use random waypoint which is an unrealistic approach due to its probabilistic nature. We have considered three other mobility models along with random waypoint model for a realistic behavior. We have considered variation in node density and node velocity. Studies are carried out in GloMoSim simulator which provides a scalable simulation environment for wireless networks.

2 Routing Protocols

The Ad hoc routing protocols, which we tested in our simulation experiments covers all the four category viz. Proactive, Reactive, Hybrid and Geographic routing. The functionality of each protocol is discussed below.

2.1 DSR (Dynamic Source Routing) [1]

It is a reactive protocol. It is designed specifically for use in multi-hop wireless networks. It comprises two mechanisms namely Route maintenance and Route discovery. These two processes work together, to allow nodes to discover and maintain routes to destinations.

Route Discovery: It is the mechanism by which a source node S which wants to send a data packet to destination D, requests for and obtains a route to D. It happens only when route to D is unknown to S.Route Maintenance: It is the mechanism by which source node S is able to detect if there is a topology change, which results in route breakage to destination. When unavailability of a route is shown, the source S can either attempt to use any other route to D which is known by S or can revoke route discovery again to find a new route.

2.2 LAR (Location Aided Routing) [2]

It is a geographic / location based routing protocol. It was built on the notion of reducing the control message overhead of Ad hoc on demand distance vector (AODV) routing protocol and Dynamic Source Routing protocol (DSR). To achieve this, source node S floods packet selectively to only a portion of the network (called Request Zone) which is likely to have the route to destination D. To identify such location, it utilizes Global Positioning System (GPS). LAR assumes that:

1. Every node in the network knows the speed of corresponding destination node.
2. Every node in the network knows the location of corresponding destination node.

It was proposed by Y.B. Ko et.al [6]. Two concepts were proposed, namely LAR Box (LAR 1) and LAR Step (LAR 2) to find if a node is member of request zone or not.

1. LAR Box Protocol: It takes into account the location of S and expected Zone for D, to check if a neighbor is within the request zone. To calculate the Expected Zone following information is used:

    a. The most recent location on D i.e. (Xd, Yd).
    b. The time at which the above location is obtained i.e. t0.
    c. The average velocity of D i.e. Vavg.
    d. The current time t1.

With the help of these information, an expected circle with radius R = Vavg * (t1-t0), centered on (XD, YD) is obtained. The Request Zone is a rectangle area with Source S in one comer (Xs, Ys), and the Expected Zone containing D in the other corner. If a neighbor finds that, it is within the Request Zone, it forwards the route request packet further in the network. A node which is not a neighbor, knows that it is within the Request Zone by using the location of the neighbor (that sent the route request packet) and the Expected Zone for D.

2. LAR Step Protocol: It takes into account the distance. Suppose that the distance between current node S and D is x and distance between neighbor I (which sent the route request packet) and D is y. Now if x<y, then current node falls within the request zone. Hence it can forward the route request packet further.

2.3 OLSR (Optimized Link State Routing) [3]

Optimized Link state Routing protocol is a proactive routing protocol; hence the routes are always immediately available when needed. OLSR is an optimization version of a pure link state protocol. The change in topology causes flooding of information to all nodes. In order to reduce this overhead, Multi point relays (MPR) are used. MPR is used to:

1. Reduce the flooding by minimizing the same broadcast in a particular region.
2. Provide the shortest path.

For efficient routing four types of control messages are used:

a) Hello message is sent periodically to all neighbors node having node's identifier, list of node's neighbors, its MPRs and its neighbors. This is done to find updates about the link status and host's neighbor.
b) Topology Control (TC) massage is periodically sent by a node having a set of bidirectional links between the node and a subset of node's neighbors. Its purpose is to broadcast information about one's own neighbor. TC message is broadcasted periodically, with only MPR hosts forwarding it.

c) Multiple Interface Declaration (MID) message, is used to inform other nodes that the announcing host can have multiple OLSR interface. It is broadcasted only by MPR.
d) Host and Network Association (HNA) message, is used to give information regarding external routing. The HNA message contains vital information regarding the network and the net mask addresses. This information is used to announce that a particular node can act as a gateway.

2.4 ZRP (Zone Routing Protocol) [4]

It is a hybrid and hierarchical protocol, which means that it takes advantage of both reactive and proactive routing. It works on the principle of separation of nodes and local neighborhood from global topology of entire network. The above mentioned local neighborhoods are called zones. Important assumptions are:

1. Each node may be within a single zone or multiple overlapping zones.
2. Each zone may be of diverse size.
3. The routing zone has a radius ρ (in hops).

The nodes of a zone are divided into 2 parts:

a) Peripheral nodes: Nodes which are equidistant to central node, with same radius ρ.
b) Interior nodes: Nodes which are non-equidistant to central node, with radius less than ρ.

In order to control the number of nodes in the routing zone, transmission power of the nodes is adjusted i.e. by lowering the power; number of nodes within direct reach is reduced.

ZRP utilizes two components:

1. At local level, IntrA-zone Routing Protocol (IARP) for Proactive routing component.
2. At global level, IntEr-zone Routing Protocol (IERP) for reactive routing component.

In ZRP, border casting is used instead of broadcasting. In order to direct queries to the border of the zone, the topology information provided by IARP is used. The Border cast Resolution Protocol (BRP) provides the packet delivery service in case of a border cast.

3 Mobility Models

3.1 RWP (Random Way Point mobility model)

It was proposed by Johnson and Maltz [5]. Due to its simplicity and availability at a wider scale, it took no time to become benchmark model. At the start of the simulation, each mobile node randomly selects one location in the simulation area as the destination and starts moving towards that destination with constant velocity which is chosen to be uniform and random between 0 to Vmax, where Vmax is the maximum velocity for a node. The velocity and direction of a node are independent of other nodes. When the destination is reached, the node rests for some time, called pause time Tpause. After this time elapse, again some other random destination is chosen and the process continues till the simulation ends. If Tpause = 0, then there is no rest time and the mobility becomes continuous. The pause time Tpause, is chosen between [Tpausemin, Tpausemax].

3.2 GM (Gauss Markov model)

It was introduced by Liang and Haas [6]. Here the velocity of a node is assumed to be correlated over time and is modeled as a Gauss-Markov stochastic process. Using a stochastic formula, the path is determined. When the node travels beyond the boundaries of simulation area, the direction of movement is forced to flip 180 degrees. In this manner, the boundary of simulation field is never touched. The velocity Vt at time t is dependent on velocity Vt-1 at time t-1. Hence temporal dependent property is exhibited. This degree of dependency is determined by parameter α, which reflects the randomness of Gauss-Markov process i.e. if α=0 a total randomness is observed and if α=1 a total linear motion is observed. Hence we choose α between 0 to 1according to the desired randomness. Initially, this model was used for the simulation of Personal Communication Service. But later on its use was extended to MANETs as well.

3.3 RPGM (Reference point group mobility model)

It was proposed by Hong et al. [7] in 1999. It was based on the thinking that nodes in MANET always tend to co-ordinate their movement. It takes care of the random movement of group and individual random movement of each node inside the group. Each group has a center, considered as group leader. The rest constitutes the group member.

Vtgroup represents the motion vector of group leader at time t. It defines the motion of group leader as well as the general motion trend of whole group. Because of this, the movement of group members is significantly affected. Mobility for each node is assigned with a reference point, which follows the group movement. Based on this reference point, each mobile node can be placed randomly in the neighborhood.

3.4 MG (Manhattan Grid model)

It was proposed by Bai et al. [8] in 2003. It models the mobility of nodes on streets defined by maps. The map consists of horizontal and vertical streets. Each street has two lanes for bidirectional moment. The mobile node is free to move along the horizontal and vertical lines in the grid. At the intersection of horizontal and Vertical Street, the mobile node can turn left or right or can go street. The choice is probabilistic i.e. turning left and right has a probability of 0.25 while probability for going straight is 0.5. Velocity at any particular time t is dependent on velocity at previous time t-1 and also on the velocity of node ahead on the same lane of the street.

4 State of the Art

Looking at the available literature, it is found that there are many studies done on MANET earlier. Some of them are very detailed while others are more technical. In late 90's researchers started working on MANETs, as these networks were thought of getting more useful in the future.

A very initial study was done by Das et al. [9] in 1998. They routing protocols were evaluated at packet level. The simulator used for this purpose was MaRS (Maryland Packet Simulator). They compared AODV, DSR, TORA, DSDV, EXBF and SPF. The observation was that although the routing load was lowered in new protocols, the link state and distance vector protocols gave better performance in terms of packet delivery and end to end delay. The work was extended for more cases [10] of node speed and density in 2000 with same set of protocols.

The same year authors [11] compared AODV, DSDV, DSR and TORA routing protocols on NS2 (Network Simulator 2). Random Way Point mobility model (RWP) was used for movement of nodes. For the above mentioned algorithms, the authors evaluated packet delivery ratio, routing overhead and path optimality. The node density was fixed at 50.

An important work was done by X Hong et al. [12] in 1999. They presented a survey of mobility models in cellular and multi-hop networks. They showed that group motion occurs frequently in ad hoc environments, and based on this designed a group mobility model called RPGM (Reference Point Group Mobility model). They also showed that by changing the value of parameters in RPGM, many other mobility models can be modelled. They applied this mobility model to study the behavior on clustering and routing. The simulator used was parallel simulation language Maisie. The node density was fixed at 100 and protocols compared were DSDV, AODV and HSR. For the first time they showed that performance of routing protocol depends on the choice of mobility model.

In 2000, Lee et al. [13] compared the performance of multicast protocols in Ad hoc environment. The protocols comprised of AMROUTE, FLOODING, ODMRP, CAMP and AMRIS. The metrics obtained were packet delivery ratio, number of data packets transmitted per data packet delivered, number of control bytes transmitted per data bytes delivered and number of control and data packets transmitted per data packet delivered. The number of multicast nodes was 20 with speed from 0 kmph to 72 kmph. The simulator chosen was GloMoSim. Authors concluded that mesh based protocols outperforms tree based protocols.

Authors in [14] proposed two multi path techniques for DSR protocol. It utilizes disjoint paths. For simulation MARS simulator was used and node density was fixed at 60. The mobility model was designed based on some pre-defined distribution. Performance metrics for simulation included fractions of packets dropped, end to end delay, number of route discoveries and routing load. The authors concluded that multipath routing is better than single path routing and if all the intermediate nodes are provided shortest paths, then the performance is slightly better than providing only source with alternate paths.

In 2002, a comparative study of CBR and TCP performance on OLSR and AODV protocols was done by T. Clausen et al. [15]. The variation was done for traffic, density and mobility. The common used traffic type for MANET is CBR, but the internet uses TCP. For a heterogeneous environment consisting of both, what will be the effect of TCP and CBR? Which will be preferred? The number of nodes was fixed at 50, and random waypoint mobility model was used. The simulator used was NS2. The metrics studied were control traffic overhead, delivery ratio, path length, delay, total transfer time and normalized routing load. The conclusion from the paper was, the protocols may perform comparatively when exposed to CBR, but when the same scenario is exposed to TCP, it significantly affects performance.

Barrett et al. [16] conducted a comparative analysis of IEEE 802.11, CSMA and MACA media access protocols. They considered only static ad hoc networks. The GloMoSim simulator was used to obtain number of received packets, average latency of each packet, long term fairness and throughput. They concluded that typically, all protocols degrade significantly at higher packet injection rate. Also, it happens rather sharply.

To analyze the impact of mobility on performance of routing protocols for ad hoc networks, a framework named IMPORTANT was proposed by F. Bai et al. [8]. The mobility models used were RWP, RPGM, Freeway mobility and Manhattan mobility model. The density of nodes was fixed at 40. NS 2 was used for simulation. The routing protocols considered were DSR, AODV and DSDV. The authors showed that performance of protocol shows drastic variations across mobility models. So the performance rankings of protocols will change with a change in mobility model.

An energy based performance comparison of AODV, DSR, TORA and DSDV was done by B. Chen et al. [17]. The mobility models employed were Random Waypoint, RPGM and Manhattan grid model. The simulator used was NS 2 and node density was fixed at 50. The authors concluded that reactive protocols are more sensitive to speed than proactive protocols. It is more challenging to route packets over Manhattan grid model over the others. For group movement reactive protocols are better than proactive protocols.

T. Kunz [18] provided an in depth study of one to one and many to many communication in MANET. The protocols studied were unicast routing protocol (DSR and AODV), Multicast routing protocol (ADMR, ODMRP and Extension of AODV) and Broadcast protocols (FLOOD and BCAST). Simulations were conducted on NS2 and number of nodes was fixed at 50. The performance metrics included packet delivery ratio and latency. The authors concluded that broadcast protocols, in particular BCAST perform well and that too without a high overhead.

A multilayer analysis of the influence of mobility models on AODV protocol was done by Gomez et al. [19]. The traffic flow was considered to be TCP. The performance analysis was done at three layers viz. physical layer, network layer and transport layer. The mobility models considered were RWP, Gauss Markov model, Manhattan Grid and RPGM model. The simulator used was NS2 and density of nodes was fixed at 20. The authors concluded that higher speeds does not necessarily means lower throughput.

To implement the ant mobility model which is based on the actual movement of a group of ants, simulations were conducted by Liao et al. [20].The effect of this mobility model on DSDV, DSR and AODV is examined. For the worthiness of the model ant mobility is compared to random waypoint model for same set of protocols. The simulator used is NS2. The number of nodes for the simulation is 50. The metrics evaluated were packet throughput ratio, average end to end delay and normalized routing load. The authors concluded that trace models like ant mobility are more accurate than synthetic models like random waypoint mobility. But this accuracy comes at a cost of difficult and time consuming process.

Atsan et al. [21] classified and compared the performance of mobility model for MANET protocols. The protocol studied was AODV and simulator chosen was SWANS. Four mobility models were considered viz. random direction, boundless simulation area model, random walk and random waypoint model. The metrics considered were average message activity, average route request completion rate and average RREQ message sent per route added. The density of node was fixed at 50. Authors concluded that although RAP does not give the best performance for all the used performance metrics, it is most consistent for varying simulation levels.

A realistic simulation based study of MANET protocols was made by Marinoni et al. [22]. They proposed a new and realistic Urban Mobility Model (UMM), which models realistic user motion and signal propagation in a city like scenario. The mobility models namely RWP, UMMoff (UMM with radio constraints activated) and UMMon (UMM with radio constraints deactivated) were applied on DSR protocol. The number of sender/receiver was 20 pairs for all experiments. NS2 was used for simulation. The metrics calculated were packet delivery ratio, end to end delay, path length and routing overhead. The authors concluded that trivial RWP is too simplistic and too narrow in its scope. Hence a realistic model like UMM can be a better choice.

Pirzada et al. [23] compared performance of multi-path AODV and DSR protocols in hybrid mesh networks.NS2 was the preferred simulator. The number of mesh clients was fixed at 50 and number of mesh routers was fixed at 16. Random waypoint model was considered for mobility. Packet loss, aggregate good put, packet delivery percentage, routing packet overhead, average latency and path optimality were the metrics calculated. The authors concluded that mesh networks with inclusion of mesh router gives better performance.

G. Jayakumar et al. [24, 25] compared performance of DSR and AODV for random waypoint and Manhattan grid model. The node density was fixed at 20 nodes. The performance metrics included packet delivery fraction, average end to end delay, normalized routing load and normalized mac load. The simulator used was NS2. The authors observed a very clear trend between mobility metric, connectivity and performance.

In 2009 Karthikeyan et al. [26] studied the performance of broadcasting methods in MANET. The techniques employed for broadcasting was simple flooding and probability based flooding. The simulations were performed on NS2. The number of mobile nodes was fixed at 24. The performance metric included normalized routing load for DSDV protocol. The authors concluded that probabilistic broadcast performs better than simple flooding.

A comparative performance analysis of DSDV, AODV and DSR routing protocols was done by Tuteja et al. [27]. For simulation, NS2 was used. The metrics included packet delivery ratio, throughput, end to end delay and routing overhead. 25 nodes were considered for simulation. Random waypoint model was used to define movements of node. The authors concluded that with the increase in mobility of nodes performance degrades irrespective of the choice of three discussed protocols.

Unicast and broadcast routing protocols of MANET were evaluated by Debnath et al. [28]. Both one to one and many to many communications were addressed in detail. DSR and BCAST protocol were simulated on NS2. The number of nodes was fixed at 50. Mobility model used was random waypoint mobility model. The performance metrics included packet delivery ratio, packet latency, normalized routing load, normalized mac load and throughput. The authors concluded that BCAST protocol works well in most scenarios and is robust even with high traffic environments.

Barakovic et al. [29] compared the performance of MANET routing protocols AODV, DSR and DSDV. Simulations were carried on NS2. Packet delivery ratio, average end to end delay and normalized routing load were the performance metrics. The numbers of source nodes varied from 10 to 30. The mobility of nodes was defined by random waypoint model. The conclusion form the study was that all the protocols reacted in similar ways for low mobility and low load conditions, while DSR outperformed AODV and DSDV with increasing mobility and load.

In 2011, Mohapatra et al. [30] studied the effect of change in network size, mobility and pause time on AODV, OLSR and DSDV. The number of nodes was fixed at 30. The choice of simulator was NS2 and that of mobility model was random waypoint model. Throughput, Routing overhead, delay and packet delivery ratio were calculated for varying number of nodes, varying pause time and varying network area. The authors concluded that for highly mobile random network OLSR is preferred.

Performance comparison of relatively newer set of protocols viz. LANMAR, LAR1, DYMO and ZRP was done by Singh et al. [31]. Qualnet simulator was chosen for the experiments. 50 nodes were considered for the scenario. Random waypoint model was used to define the mobility pattern of nodes. The performance metrics were average end to end delay, packet delivery ratio, throughput and average jitter. The authors concluded that LANMAR is the best scheme in terms of end to end delay and jitter while LAR1 is best in terms of packet delivery ratio and throughput.

A comparative study was done by Saada et al. [32] to evaluate the performance of protocols. GloMoSim simulator was used for experiments. To compare the performance DSDV, AODV, ARPM and SHARP protocols were considered. The number of nodes was different for different scenarios. For static scenario, it was fixed at 70 while for dynamic scenario it varied from 10 to 140. Random waypoint model was the mobility model and the metrics included overhead, route discovery delay and throughput. The conclusion derived from the work was that DSDV is better for small networks and AODV is better for large networks.

From the above given analysis, we conclude that although a lot of comparative studies have been carried out on MANET routing protocols based on one or more mobility models, most of them have relied on Random waypoint model, which due to its probabilistic nature is unrealistic. Most of the work has been done considering either variation in node density or node speed along with mobility models. We have considered variation in speed and density of nodes together as a parameter to study the effects on a wider perspective. Also for the choice of mobility models we have considered RWP, MG, GM and RPGM. We have taken a candidate protocol from each group of protocols viz. reactive, proactive, location based and geographic.

5 Simulation Setup

To study the performance of routing protocols we evaluated throughput, end to end delay and ratio of packet loss. The metrics are described as follows.

- Throughput: It is the ratio of number of packets received at destination to the number of packets originated at source. The source follows CBR (Constant bit rate) traffic. It depicts the loss rate.
  Throughput = Data packets received / Data packets sent
- End to end delay: It is the average amount of time that is taken by a packet to reach final destination from source. It includes the route discovery wait time, which a node may experience in case a route is not available.
  Average delay = Σ (tr - ts) /Pr, where ts is the packet send time and tr is the packet receive time.
- Packet loss: It is the fraction of packet lost on their route to destination. The loss is usually due to congestion on the network and buffer overflows.
  Packet loss = Number of lost packets / number of received packets

To generate mobility patterns for MG, RWP, GM and RPGM Bonn Motion tool is used. We have studied the impact of speed and node density on performance of the network. To compare the protocols, same set of scenarios is utilized for each one. The simulator used is GloMoSim [33]. The simulation parameters are given below.

Parameter Name	Value
Speed of node	0 to 20 m/s
Density of node	5 to 200
Number of CBR sources	10
Speed of CBR link	10 packets per second
Packet Size	512 bytes
Wireless Radio	802.11
Transmission Range	50 m
Transmission rate	1 Mbps
Area of simulation	1500m x 1500m
Simulation time	300 seconds

The parameters chosen for mobility models are as follows:

Model	Parameter	Value(s)
RWP (Random Way Point)	Pause time	0 sec
	Min. speed	0 m/s
	Max. Speed	20 m/s
MG (Manhattan Grid)	No. of blocks along y-axis	2
	No. of blocks along x-axis	10
	Min. Speed	0 m/s
	Max. Speed	20 m/s
	Probability of going straight	0.5
	Probability of going right	0.25
	Probability of going left	0.25
RPGM (Reference Point Group Mobility)	Average no. of nodes per group	5
	Max. distance to center of group	5 m
	Min. Speed	0 m/s
	Max. Speed	20 m/s
GM (Gauss Markov)	Min. Speed	0 m/s
	Max. Speed	20 m/s

6 Results and Discussion

6.1 Throughput

LAR, OLSR, DSR and ZRP were tested under RWP, MG, RPGM and GM models. We varied the speed of the nodes from 0 to 20 m/s at interval of 5. The node density was varied from 50 to 200 in the intervals of 50. From the results, it is clear that LAR and OLSR outperform others in terms of throughput. The results are given in Fig. 1 to 4. Throughput metrics is almost equal to 100 % for OLSR and LAR. But, in the case of random waypoint model although OLSR still outperforms others but the throughput is reduced largely. This is due to the fact that in random waypoint model link breakage is more often for higher speeds and hence the throughput decreases for almost all protocols. At higher speeds i.e. 10 to 20 m/s LAR and OLSR are the preferred choice for better throughput. But, at lower speeds the case changes, LAR behaves better than others for RWP and RPGM. Also, the applications which utilize RWP should use LAR for lower speeds. While the applications, which uses other mobility models can either opt for OLSR or LAR for all cases of mobility and speed.

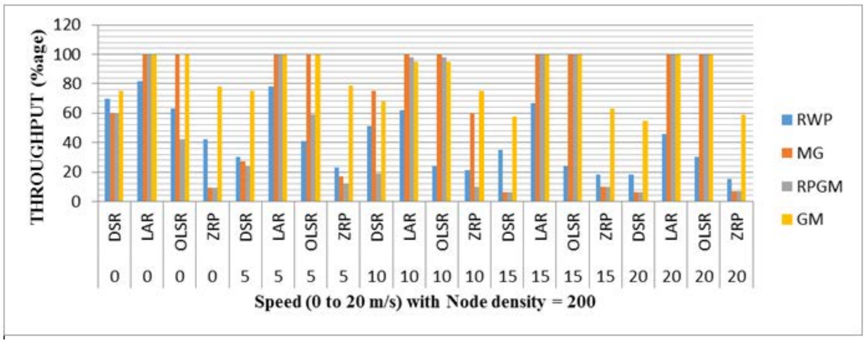

Figure 1: Throughput at Node Density 200, with speed varying from 0 to 20 m/s

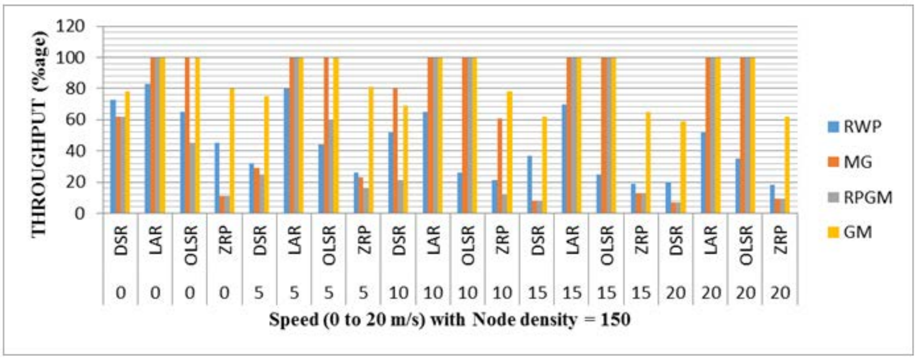

Figure 2: Throughput at Node Density 150, with speed varying from 0 to 20 m/s

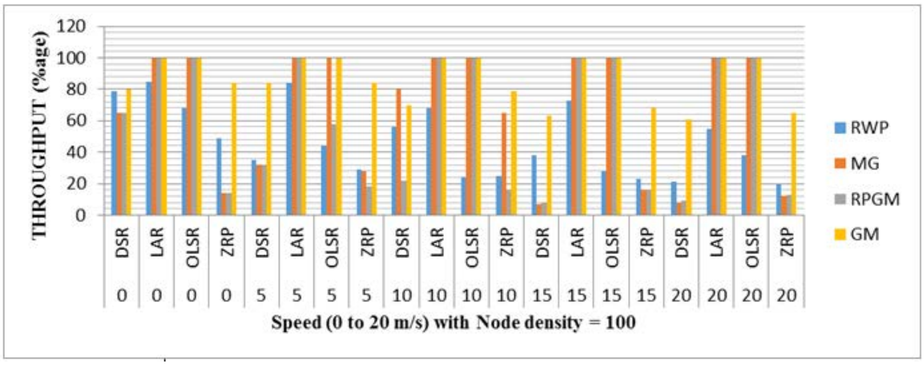

Figure 3: Throughput at Node Density 100, with speed varying from 0 to 20 m/s

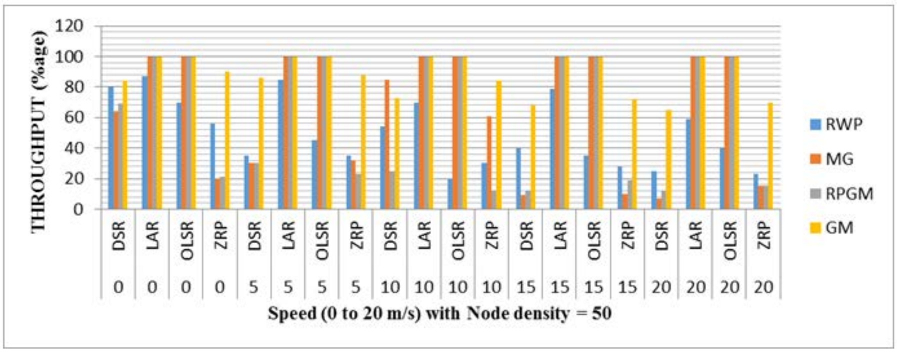

Figure 4: Throughput at Node Density 50, with speed varying from 0 to 20 m/s

6.2 End to End Delay

The results for end to end delay are shown in Fig. 5 to 8. The values for delay for some case is very small, and to make that portion visible additional sub graph is given, which highlights the smaller values. These sub graphs are numbered from 5a to 8a. The speed of the nodes is varied from 0 to 20 m/s in steps of 5 and node density is varied from 50 to 200 in steps of 50. LAR exhibits lowest end to end delay for almost all the cases of speed and node density compared to other protocols. With increase in speed and number of nodes the delay also increases. It happens because at higher speeds, connectivity decreases and hence accounts for higher delays. The applications which require less end to end delay should use LAR, as is clear from the above discussion.

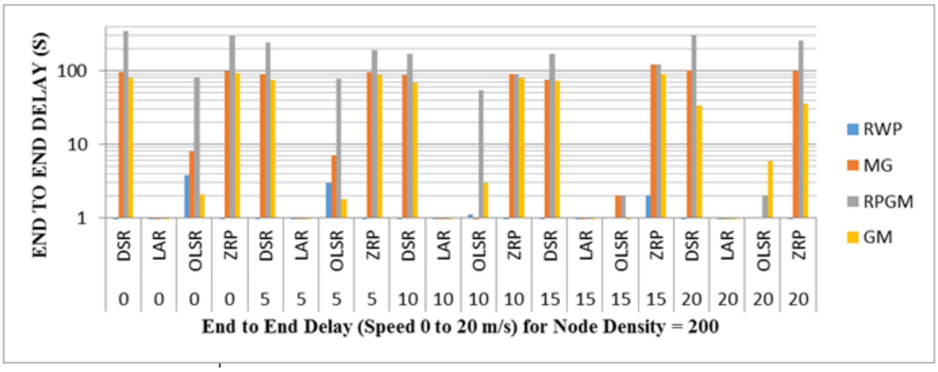

Figure 5: End to End Delay at Node Density 200, with speed varying from 0 to 20 m/s

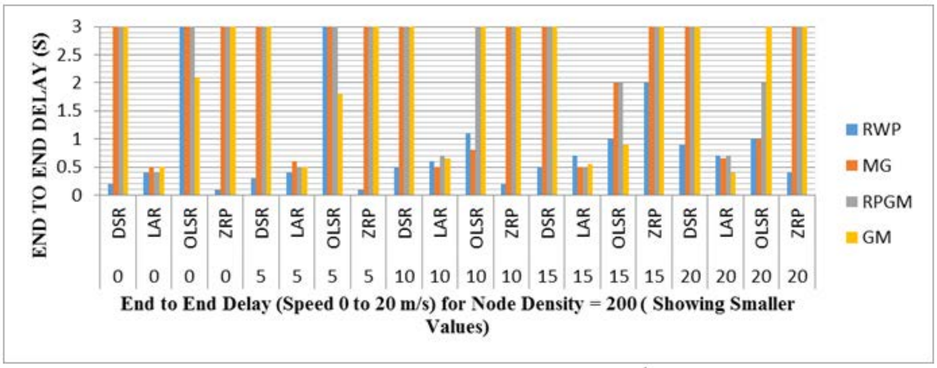

Figure 5a: End to End Delay at Node Density 200, speed varying from 0 to 20 m/s (Only smaller values are shown)

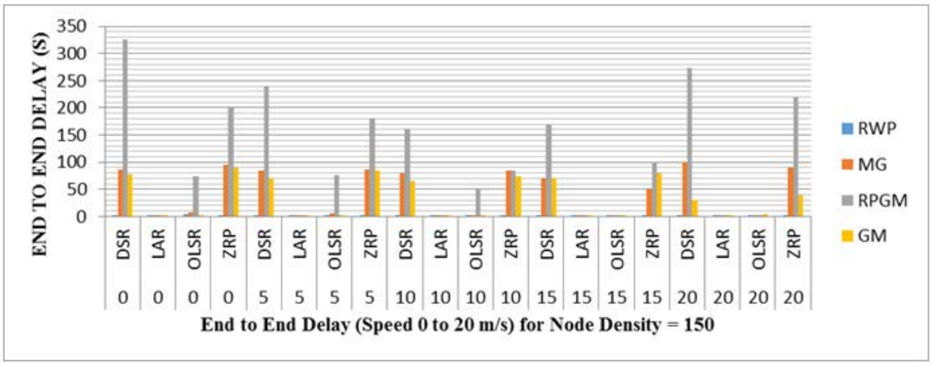

Figure 6: End to End Delay at Node Density 150, with speed varying from 0 to 20 m/s

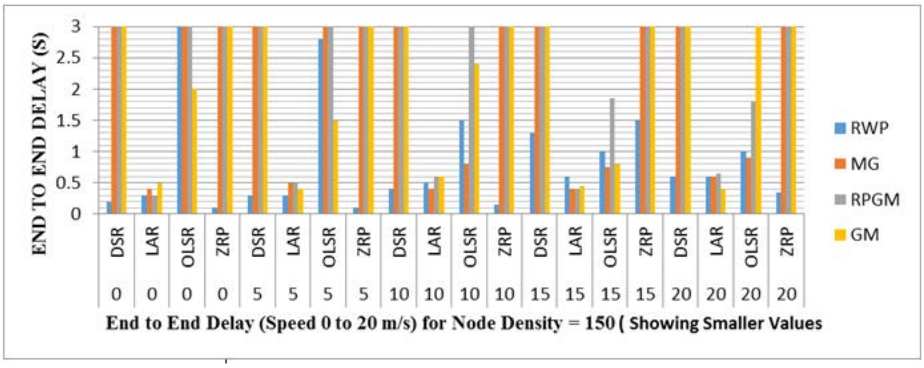

Figure 6a: End to End Delay at Node Density 150, speed varying from 0 to 20 m/s (Only smaller values are shown)

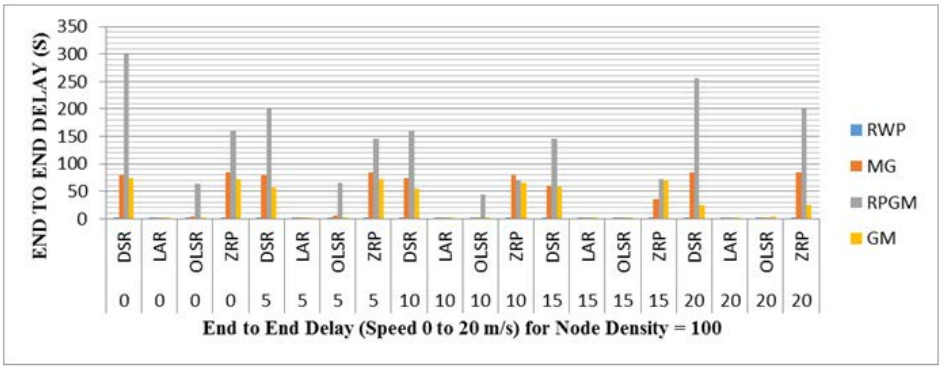

Figure 7: End to End Delay at Node Density 100, with speed varying from 0 to 20 m/s

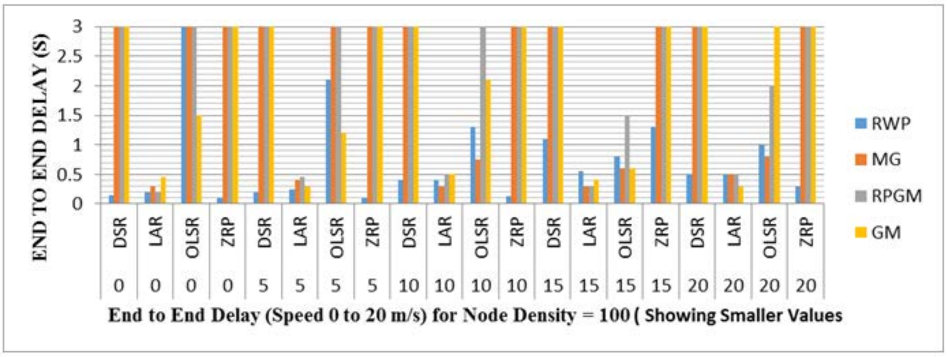

Figure 7a: End to End Delay at Node Density 100, speed varying from 0 to 20 m/s (Only smaller values are shown)

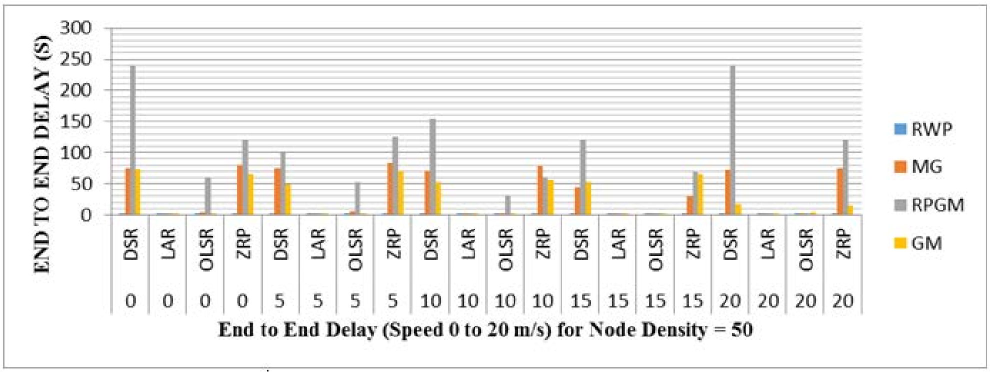

Figure 8: End to End Delay at Node Density 50, with speed varying from 0 to 20 m/s

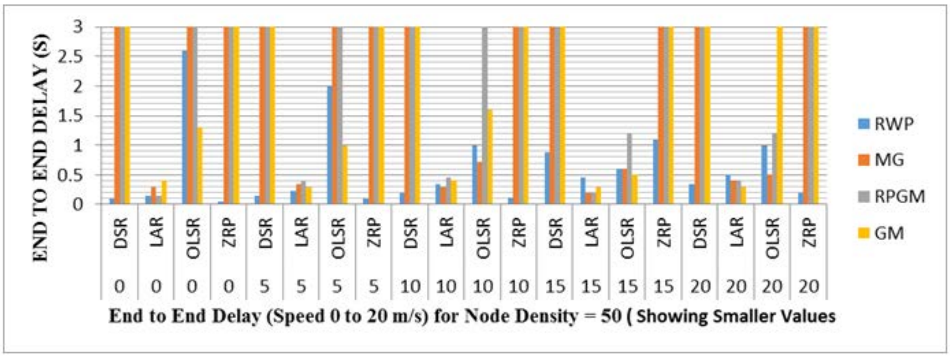

Figure 8a: End to End Delay at Node Density 50, speed varying from 0 to 20 m/s (Only smaller values are shown

6.3 Packet Loss

The results for packet loss are shown in fig 9 to 12. The speed of the nodes is varied from 0 to 20 m/s in steps of 5 and node density is varied from 50 to 200 in steps of 50. At all node density, Gauss

Markov model gives the lowest packet loss for all protocols, specially LAR and OLSR. This can be due to the fact that the value of speed and direction at the nth location is dependent on the previous value and a random variable. It means that probability of a node to remain in its old entirety is more, hence incurring low packet loss. LAR has the least packet loss with respect to other protocols for almost all cases of node speed and density except random way point model. With increase in speed the probability of packet loss also increases. It happens because at higher speeds, connectivity decreases and hence accounts for packet loss delays. The applications which require less packet loss should use LAR, as is evident from the results.shown)

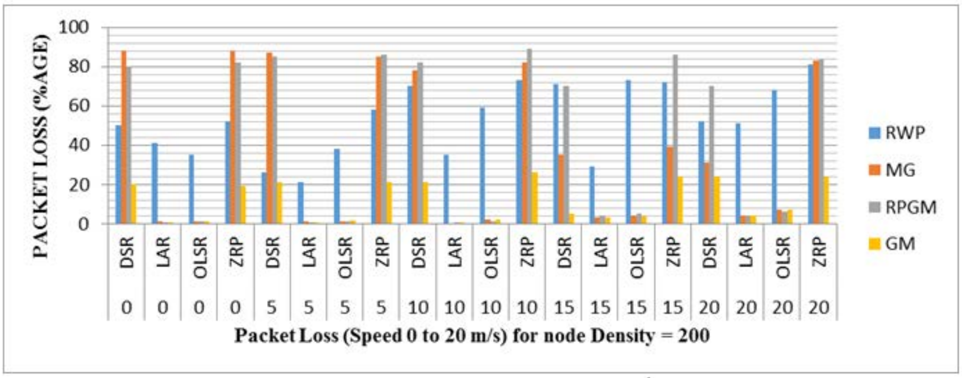

Figure 9: Packet Loss at Node Density 200, with speed varying from 0 to 20 m/s

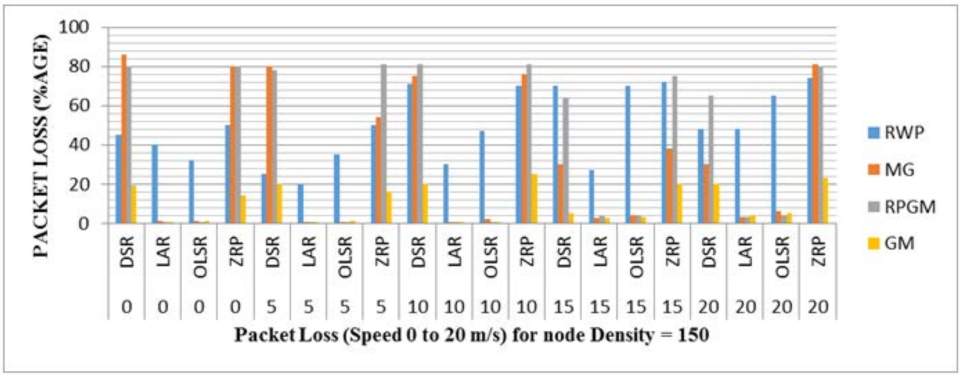

Figure 10: Packet Loss at Node Density 150, with speed varying from 0 to 20 m/s

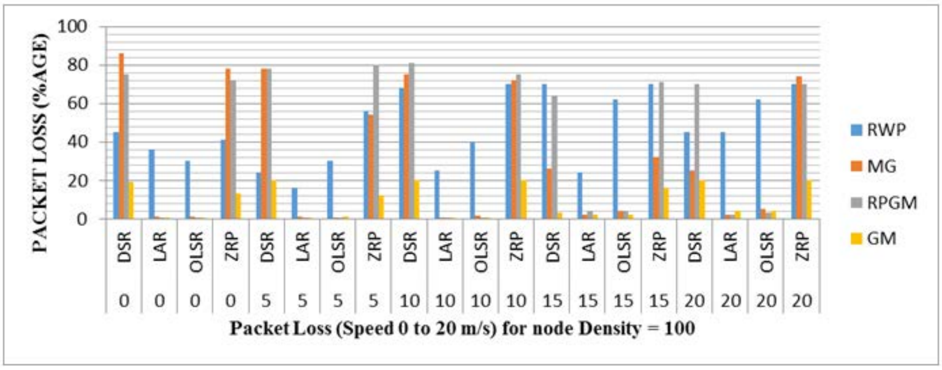

Figure 11: Packet Loss at Node Density 100, with speed varying from 0 to 20 m/s

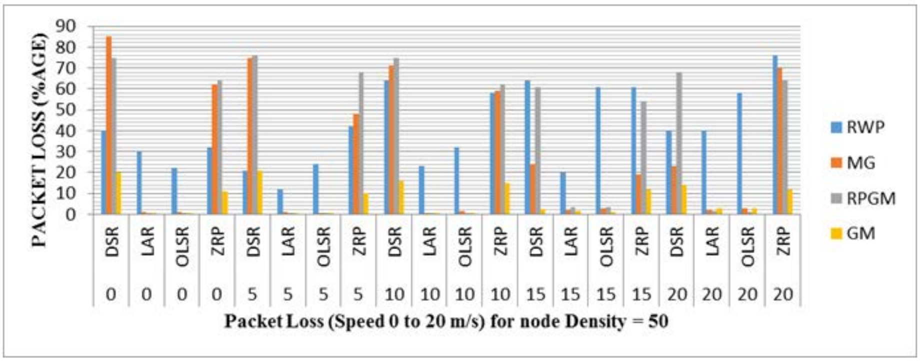

Figure 12: Packet Loss at Node Density 50, with speed varying from 0 to 20 m/s

7 Conclusions

In this work we presented a simulation based performance analysis of MANET routing protocols. From the reactive, proactive, geographic and location based protocols, one candidate protocol was chosen for analysis. The mobility models used were random way point, gauss Markov, Manhattan Grid and reference point group mobility model. The variation in node speed was done from 0 to 20 m/s and node density from 50 to 200. The analysis of the throughput suggests that LAR and OLSR with MG model gives 100 %, due to the fact that a restriction in mobility area in a grid betters the throughput. LAR and OLSR give best performance in RPGM, due to the presence of a group leader, who is responsible for the mobility of the group and even distribution of group members. Due to localization property LAR performs better under RWP. This is due to the fact that in RWP model the nodes are distributed such that they are able to move freely and independently of others.

REFERENCES

[1] D. Johnson, Y. Hu, D. Maltz, "The Dynamic Source Routing Protocol (DSR) for mobile Ad Hoc Networks for IPV4", rfc4728.

[2] Young-Bae Ko, Nitin H. Vaidya, "Location-Aided Routing (LAR) in mobile ad hoc networks", Wireless Networks, Volume 6, Issue 4, pp 307-321.

[3] T. Clausen, P. Jacquet, "Optimized Link State Protocol (OLSR)", rfc3626.

[4] Zygmunt J. Haas, Marc R. Pearlman, Prince Samar, "The Zone Routing Protocol for Adhoc networks", draft-ietf-manet-zone-zrp-04.txt, july2002

[5] J. Broch, D. A. Maltz, D. B. Johnson, Y.C. Hu, and J. Jetcheva, "A performance comparison of multi-hop wireless ad hoc network routing protocols", in Proceedings of the Fourth Annual ACM/IEEE International Conference on Mobile Computing and Networking(Mobicom98), ACM, October 1998.

[6] B. Liang, Z. J. Haas, "Predictive Distance-Based Mobility Management for PCS Networks", in Proceedings of IEEE Information Communications Conference (INFOCOM 1999), Apr. 1999.

[7] X. Hong, M. Gerla, G. Pei, and C.C. Chiang, "A group mobility model for ad hoc wireless networks", in Proceedings of ACM International Workshop on Modeling, Analysis, and Simulation of Wireless and Mobile Systems (MSWiM), August 1999.

[8] F. Bai, N. Sadagopan, and A. Helmy, "Important: a framework to systematically analyze the impact of mobility on performance of routing protocols for ad hoc networks", in Proceedings of IEEE Information Communications Conference (INFOCOM 2003), San Francisco, Apr. 2003.

[9] 10. Samir R. Das, Robert Castaneda, Jiangtao Yan, and Rimli Sengupta, "Comparative Performance Evaluation of Routing protocols for Mobile, Ad hoc Networks", in Proceedings of Seventh international Conference on Computer Communication and Networks (IC3N), Lafayette, LA, October 1998, pp. 153-161.

[10] Samir R. Das, Robert Castaneda, and Jiangtao Yan, "Simulation-based performance evaluation of routing protocols for mobile ad hoc networks", Journal of Mobile Networks and Applications, Volume 5, pp. 179–189, 2000.

[11] Josh Broch, David A. Maltz, David B. Johnson, Yih-Chun Hu, and Jorjeta Jetcheva, "A Performance Comparison of Multi-Hop Wireless Ad Hoc Network Routing Protocols", in Proceedings of the Fourth Annual ACM/IEEE International Conference on Mobile Computing and Networking (MobiCom'98), October 25–30, 1998, Dallas, Texas, USA.

[12] Xiaoyan Hong, Mario Gerla, Guangyu Pei, Ching-Chuan Chiang, "A group mobility model for ad hoc wireless networks", in Proceeding of the 2nd ACM international workshop on Modeling, analysis and simulation of wireless and mobile systems, MSWiM '99, ACM New York, NY, USA, pp. 53-60.

[13] Sung-Ju Lee, W. Su, J. Hsu, M. Gerla, R. Bagrodia, "A performance comparison study of ad hoc wireless multicast protocols" in proceedings of Nineteenth Annual Joint Conference of the IEEE Computer and Communications Societies, INFOCOM 2000, pp. 565-574.

[14] Asis Nasipuri, Robert Castaneda, Samir R. Das, "Performance of Multipath Routing for On-Demand Protocols in Mobile Ad Hoc Networks", Journal of Mobile Networks and Applications, Volume 6, pp. 339–349, 2001,Kluwer Academic Publishers.

[15] Thomas Clausen, Philippe Jacquet, and Laurent Viennot, "Comparative study of CBR and TCP performance of MANET routing protocols", in Workshop on Broadband Access, INRIA Lab, France, Sep. 2002.

[16] C. L. Barrett, M. Drozda, and M.V. Marathe, "A comparative experimental study of media access protocols for wireless radio networks", Wireless Communications and Networking Conference (WCNC 2002), 17-21 Mar 2002, pp. 405 – 411.

[17] Bor-rong Chen and C. Hwa Chang, "Mobility Impact on Energy Conservation of Ad Hoc Routing Protocols", SSGRR 2003, Italy, Jul. 28-Aug. 2, 2003.

[18] Thomas Kunz, "Reliable Multicasting in MANETs", Contract Report, DRDC-Ottawa, July 2003.

[19] C. Gomez, X. Marchador, V. Gonzalez, and J. Paradells, "Multilayer analysis of the influence of mobility models on TCP flows in AODV ad-hoc networks", in Proceedings of the 14th

IEEE Workshop on Local and Metropolitan Area Networks (LANMAN), 2005, 18 Sept. 2005, pp 1-6.

[20] Hsien-Chou Liao, Yi-Wei Ting, Chia-Meng Chen, and Chou-Chen Yang, "A Performance Comparison of Ad Hoc Routing Protocols Based on Ant Mobility Model", Information Technology Journal, Vol. 4, No. 3, 2005, pp. 278-283.

[21] Emre Atsan, Oznur Ozkasap, "A Classification and Performance Comparison of Mobility Models for Ad Hoc Networks", Ad-Hoc, Mobile, and Wireless Networks, Lecture Notes in Computer Science Volume 4104, 2006, pp 444-457.

[22] Stefano Marinoni, Hannu H. Kari, "Ad hoc routing protocol's performance: a realistic simulation based study", Telecommunication Systems, December 2006, Volume 33, Issue 1-3, pp 269-289.

[23] Asad Amir Pirzada, Marius Portmannt, and Jadwiga Indulska, "Performance Comparison of Multi-Path AODV and DSR Protocols in Hybrid Mesh Networks", in Proceedings of 14th IEEE International Conference on Networks (ICON), Sept. 2006, pp. 1-6.

[24] Geetha Jayakumar, and G. Gopinath, "Performance Comparison of Two On-demand Routing Protocols for Ad-hoc Networks based on Random Way Point Mobility Model", American Journal of Applied Sciences, Vol. 5, Issue 6,2008, pp. 659-664.

[25] Geetha Jayakumar, and G. Gopinath, "Performance comparison of MANET protocols based on Manhattan Grid Mobility Model", Journal of Mobile Communication, Vol. 2, Issue 1, 2008, pp. 18-26.

[26] N.Karthikeyan, V. Palanisamy, and K. Duraiswamy, "Performance Comparison of Broadcasting methods in Mobile Ad Hoc Network", International Journal of Future Generation Communication and Networking, Vol. 2, No. 2, June 2009, pp. 47-58.

[27] Asma Tuteja, Rajneesh Gujral, and Sunil Thalia, "Comparative Performance Analysis of DSDV, AODV and DSR Routing Protocols in MANET using NS2", in Proceedings of International Conference on Advances in Computer Engineering (ACE), 20-21 June 2010, pp. 330 – 333.

[28] Sumon Kumar Debnath, Foez Ahmed, and Nayeema Islam, "Performance Evaluation of Unicast and Broadcast Mobile Ad-hoc Networks Routing Protocols", International Journal of Computer Science and Information Security (IJCSIS), Vol. 7, No. 1, 2010, pp. 40-46.

[29] Sabina Barakovi, Jasmina Barakovi, "Comparative Performance Evaluation of Mobile Ad Hoc Routing Protocols", MIPRO 2010, May 24-28, 2010, Opatija, Croatia, 518-523.

[30] S. Mohapatra, P. Kanungo, "Comparative Performance Analysis of MANET Routing Protocols Using NS2 Simulator", Computational Intelligence and Information Technology, Communications in Computer and Information Science, Volume 250, 2011, pp 731-736.

[31] Dinesh Singh, Ashish K. Maurya, Anil K. Sarje, "Comparative Performance Analysis of LANMAR, LAR1, DYMO and ZRP Routing Protocols in MANET using Random Waypoint

Mobility Model", in proceedings of 3rd International conference on Electronics Computer Technology (ICECT), 8-10 April 2011, pp. 62-66.

[32] Imad I. Saada, Raed Zaghal, "MANET Routing Protocols: Comparative Study", Journal of Al-Quds Open University for Research and Studies, Vol. 30, Part 1, June 2013, pp. 9-26.

[33] X. Zeng, R. Bagrodia, and M. Gerla, "GloMoSim A Library for Parallel Simulation of Large-Scale Wireless Networks", in Proceedings of the 12th ACM Workshop on Parallel and Distributed Simulation, Canada, 1998, pp. 154-161.

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Priority Based Data Dissemination in MANET

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ABSTRACT

Opportunistic networks can provide network services in an infrastuctureless ad-hoc manner. This is especially useful if either no infrastructure is available or an existing infrastructure is not to be used. This can be due to high cost, or security considerations. Much effort went into the development of protocols and other theoretical background over the last decade. However use cases and applications have been neglected. In this paper we present a protocol for priority based data dissemination as well as two applications built on top of the protocol. These were implemented as Android applications on top a framework for ad-hoc communication. To show the functionality and to give some performance metrics, real world evaluations were conducted.

Keywords: 802.11, data dissemination, delay-tolerant networking, dtn, dtn-applications, real-world.

1 Introduction

Despite the continuing digitalization, increasing interconnectivity and a growing availability of mobile network services, there exist plenty of scenarios where no infrastructure exists or where it is overloaded. During outdoor activities like hiking or mountaineering there often is no, or only bad data connectivity. In general rural areas with low population densities are economically not attractive for providers. But also in urban environments an infrastructure might be unavailable, for example underground or indoors.

After catastrophic events or during power outages a previously existing infrastructure might no longer be available. In this case communication has to be provided by some other means. There are also other factors when one does not want to use an existing infrastructure. Using roaming data services during inter country travel can become quite expensive. Furthermore security considerations can lead to a preference of direct communication between devices and the avoidance of potentially insecure infrastructure. Highly unstable and overloaded data services can be encountered during trade shows or festivals. In these instances channel bandwidth and medium access can no longer accommodate the demand of hundreds of users.

In these aforementioned scenarios smaller decentralized networks like MANETs [1] can help to mitigate some of the problems. Opportunistic networks transmit Messages without prior knowledge of the underlying topology, which in a mobile scenario is constantly changing. Transmitting data in such a network is possible even if there is no end-to-end connectivity between two nodes [1]. Such

[1]Mobile Adhoc Networks (MANETs) are comprised of nodes that communicate in an ad-hoc manner. As such they don't rely on preexisting infrastructure.

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networks are also called delay tolerant networks, since the delay in a partitioned network cannot be predicted a priori.

Due to the constant changes that are innate to these kind of networks, the contact time between two devices cannot be predicted and might be very short. This gave rise to the idea of a prioritized packet transfer in which”more important” data is exchanged first.

In 2012 Jörg Ott concluded [2] that there is a mismatch between the foundations of delay tolerant networking (such as protocols, models and systems) and real world applications that make use of these foundations. In this paper we present two applications for priority based data exchange in opportunistic networks and a generic protocol on which the applications are based. We aim to provide a solution to reliably distribute data throughout a network while prioritizing relevant data. While other approaches often assume unlimited storage capacity we address the limited memory problem in our protocol design.

The first section gives an overview of related work. Then our own approach is outlined including the protocol design and the packet layout. Afterwards the two applications are presented. The first being a short message service and the second implements a service for exchanging location based on OpenStreetMap data. For each application the evaluation procedure is described and the results presented. The paper finishes with a conclusion and an outlook.

2 Related Work

Over the last years many papers have been published that introduce protocols and try to improve the data dissemination rate with various means. Others deal with the correct modeling of mobility or human behavior. In this section we present approaches to opportunistic networking that are application driven and make use of some kind of prioritization.

A prioritization of data has been researched by Batabyal and Bhaumik in [3], who proposed a "Fair queue message scheduling policy". This policy prioritizes based upon the overall waiting time, which is the summarized time a packet had to wait at all previous network links before being transferred over the next hop. To minimize the variance of the waiting time at every hop the message with highest overall waiting time is preferred.

Ott et al. present a floating content concept in [4] where they couple content to an anchor zone. This coupling is combined with a time to live which limits the availability of the content. The anchor zone has a replication range within which the holder of a content item replicates it actively and an availability range which describes the area in which the content items aren’t replicated but are still present. Outside these two ranges the content is deleted. To minimize spamming of the network, content items with large anchor zones are deleted when the content holders run out of storage space. Further prioritization is for items with small storage amount which are close to their anchor.

With the so called "Drop policies" Soares et al. propose in [5] three priority classes. Prioritization takes place when too many incoming messages cause a buffer overflow and when the messages are sent. As this could lead to a monopolization of the network resources by the messages of the highest priority class the messages with the longer remaining TTL are preferably transmitted. A longer remaining TTL stands for a higher probability that the message reaches its destination.

While Batabyal and Bhaumik prioritize regarding the overall waiting time, Ott et al. choose the anchor zone, the distance to it and the storage amount as criteria. Soares et al. Order the messages by only three distinct priority classes and their TTL. These three proposals have the disadvantages that they aren't content sensitive, they don't prioritize when sending the message or they don't support a continuous prioritization. Therefore this paper proposes a priority based data dissemination with content sensitive prioritization that is used when sending and storing messages.

3 Data Dissemination Strategy

The main goal of a data dissemination strategy is to provide a data set to every device in the network - or at least in a certain "hop-distance" or geographical radius. In a mobile environment the goal is limited by certain constraints: Devices only have a finite amount of storage. Power consupmtion is an issue due to a limited supply of battery power and due to wireless communication channel bandwidth and allocation are problematic. Our approach aims to mainly address the storage problem. Nevertheless the prioritization of data also reduces the number of unnecessary packets and should therefore also reduce energy consumption and channel load. Wireless communications are inherently broadcast in nature. This circumstance is of benefit for data dissemination. Our protocol addresses every device in communication range and as such reduces redundant resends to further reduce the energy and bandwidth footprint. As a further requirement we wanted to implement a protocol that is self pruning, meaning that that "out of date packets" are no longer distributed in the network. This is mainly achieved by a dynamic prioritization function that is described in further detail in the following section.

3.1 Prioritization

As an improvement on the protocols presented in the related work section we introduce a content sensitive prioritization function. This function is not restricted to discreet values but can be implemented as a continuous function. As the priority is determined by the corresponding application, the function needs to be implemented on a per application basis. As an argument it is given an application packet and is expected to return a value denoting a priority value. Per design we chose smaller values to represent higher importance. As this function is dynamic it may produce different values per call. This makes it possible to implement "auto pruning" based on time or location of the device.

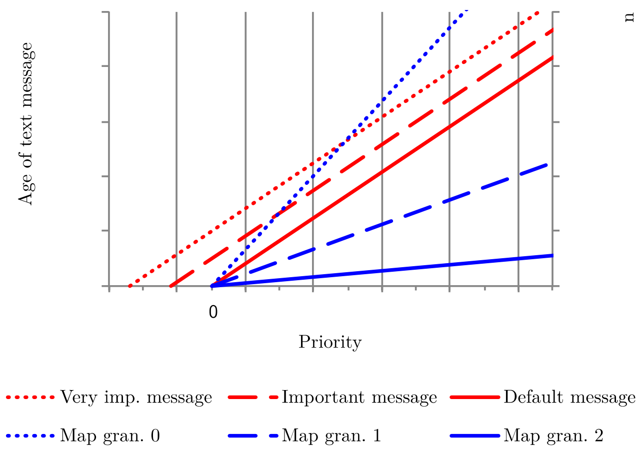

Figure 1: Graph of the priority function for text- and map-data. The lower the priority value the more important the content

As an example for such a content sensitive function, figure 1 shows the priority value of data packets generated by the applications presented in section IV. The blue data points depict the values for the map application. The red data points are for the text application. The values for the axis and different patterns are explained in the corresponding application sections.

3.2 Protocol Design

The protocol design is separated in two parts. The first part is the meta data communication. The meta data trigger the exchange of the actual payload, the data packets, between the single devices. This payload exchange forms the second part of the protocol.

3.2.1 Meta Data

The meta data describe the data that are present on a device with IDs, which are individual hash values for every data element. A meta data packet is shown in figure 2. It includes all IDs of the data sets present on a device.

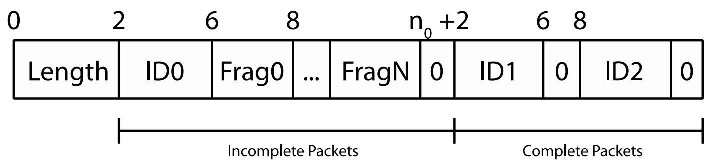

Figure 2: Packet layout of the meta data packets. After a length field the available data sets are listed. Each data set is identified by its ID and a 0-terminated list of missing fragments.

Because a data set can exceed the maximum packet length which is defined by the maximum transfer unit (MTU), the data sets can be fragmented. If a data set is fragmented and some fragments are missing, the continuous numbers of the missing fragments are listed after the data set's ID. The meta data communication is triggered by three events. The first event is the appearance of a new device in the adhoc network. All devices send periodically beacons with their MAC address. With these beacons every device creates a list of its active neighbours, which enables the easy detection of new devices that are unknown to the list.

Another event is the periodical meta data communication, which is important, because the priorities can change continuously. Therefore a new meta data communication could evoke another modified reaction to the current situation. The periodical meta data communication also increases the stability of the system.

The third event that can cause meta data communication is when a device detects that fragments of a data set are missing. After a defined waiting time interval the device with incomplete data sets sends a meta data packet to evoke the sending of the missing fragments.

When a device receives a meta data packet it first the stores the information about ids and fragments in a data structure (senderMap). It then checks if the packets are missing fragments. These are sent first if the matching data set is present to complete data sets before beginning a new transfer. Afterwards the device checks its own data sets against the ones described in the meta data packet, beginning with the one with the highest priority. If a packet is not stated in the received meta data, it is sent immediately.

To avoid continuous sending and overloading of the network, only a defined amount of data sets are sent in a row. The communication is then continued after a new meta data communication.

The following algorithm describes the meta data triggered sending of data packets in pseudo code.

Input: New meta-data packet
  $senderMap$ ←sender.ids and sender.fragments
  while incomplete datasets **do**
    SEND MISSING FRAGMENTS
  end while
  $i \leftarrow 0$
  while $priorityQueue \neq \emptyset \wedge i < 2$ **do**
    $data$ ←priorityQueue.first
    priorityQueue.removeFirst
    if data.id $\notin$ senderMap **then**
      $i \leftarrow i + 1$
      SEND DATA
    end if
  end while

3.2.2 Data Packets

The data packets carry the actual payload, which can be fragmented, as stated in section III-B1. The packet layout of a data packet is shown in figure 3. After a mode field that describes whether the data has to be stored in volatile or non volatile storage follows the individual id of this data set. The fields' number of fragments and continuous fragment number inform about the fragmentation and when a data set is complete. The last field is the payload length followed by the payload itself.

0	1	5	7	9	11	…
Mode	Id	#Fragments	Fragment#	Length	Payload	

Figure 2: Packet layout of the data packets.

The payload includes the information that is relevant for the content sensitive calculation of the priority like time stamps or geographical data.

When a data set has been sent by another device, triggered by the meta data communication, the receiving device handles every packet as follows. At first there is a check if the data set already exists. When it does and is incomplete, missing fragments are added to it. Otherwise the arriving packet is discarded. This method of fragmentation allows for packets to be reliably reassembled as long as there is no collision of ids. The integrity of the individual packets is entrusted to the underlying transport layer.

Complete data sets allow the calculation of their priority which is compared to the priority of the already present data sets on the device. These are kept in order by the priority queue. When shortage of storage occurs, the priority queue evicts the element with the lowest priority and adds the new element. If the new element has a lower priority than all other elements, it is discarded.

The following pseudo code explains the handling of new packets.

```
Input: dataPacket
  id ← dataPacket.id
  fragmentList ← dataPacket.fragments
  if ∃ Complete Dataset with id then
      DISCARD PACKET
      EXIT
  end if
  if ∃ Incomplete Dataset with id then
      if fragmentList ⊆ dataset then
          DISCARD PACKET
          EXIT
      end if
  end if
  dataset ← dataset ∪ fragmentList
  if dataset is not complete then
      EXIT
  end if
  if priorityQueue.size < MAXSIZE then
      priorityQueue ← priorityQueue + dataset
      EXIT
  end if
  priority ←PRIORITY(dataset)
  lowestPriority ← priorityQueue.lowestPriority
  if priority ≤ lowestPriority then
      EXIT
  end if
  priorityQueue ← priorityQueue − lowestPriority
  priorityQueue ← priorityQueue + dataset
```

4 Applications

4.1 Ad-hoc Framework

The ad hoc network was implemented in an ad-hoc framework developed at Ulm University called Adhoc3 which has been proposed in [6] and demonstrated at [7]. The Android software transmits and receives data packets over a wireless

802.11 ad hoc network. Using UDP broadcast packets the data is forwarded in a multi-hop fashion on the application layer. In addition to the packets needed for the actual data transmission, there is an additional type of packet sent. With these so called ”hello-packets” the clients exchange information about their current context (e.g. list of neighbours, position, etc.) and network statistics.

Within this framework the protocol for priority based data dissemination was implemented. To test and evaluate the protocol, two applications using the protocol were added to the Adhoc3 application. These will be presented in the following sections.

4.2 Text Message Service

The text message application provides an easy exchange of messages between devices in Manet. An advantage to common messenger apps is the usage without infrastructure and internet connection. Because of the possibly very short contact time between the devices, the application uses a prioritization of the messages.

The user interface of the text message service resembles the common appearance of messaging apps. As seen in figure 4 the bigger part of the space is for sent and received messages. The user can enter and send text, choose a specific user priority for this message and adjust the settings like the user name or the storage size available for the messages.

Figure 3: User interface for the text message service.

4.2.1 Linear propagation:

a) Test Setup: For the evaluation ten Android Smart¬phones with the installed alternative operating system CyanogenMod have been used. This OS provides the access of extended WiFi features that are used for the ad-hoc mode.

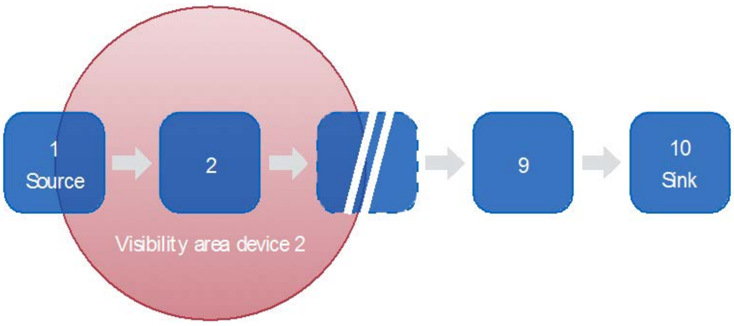

Figure 4: Test setup of the linear propagation. Every device has maximum two neighbours.

In this scenario the ten smartphones were arranged in a row to test the propagation of the text messages from the first device via the intermediate ones to the last device. The arrangement was in a way, that every device can only see its direct two neighbours like shown in figure 5.

At test start all devices were prepared to receive messages. Because there wasn't any data present only meta data packets were exchanged. In a ten seconds interval then 21 text messages were initiated on the first source device and sent to all devices in range, which was according to the test setup only the next neighbour for the source device and maximum two devices for the intermediate smartphones. Starting at the source the messages propagated autonomously from device to device until they were present on all ten smartphones.

b) Results: The averaged sending and receiving timestamps of the text messages are illustrated in figure 6. The sent messages propagated in on average 24.14 seconds from device 1 to device 10. The relatively regular propagation of the messages along the linear chain of devices can be seen.

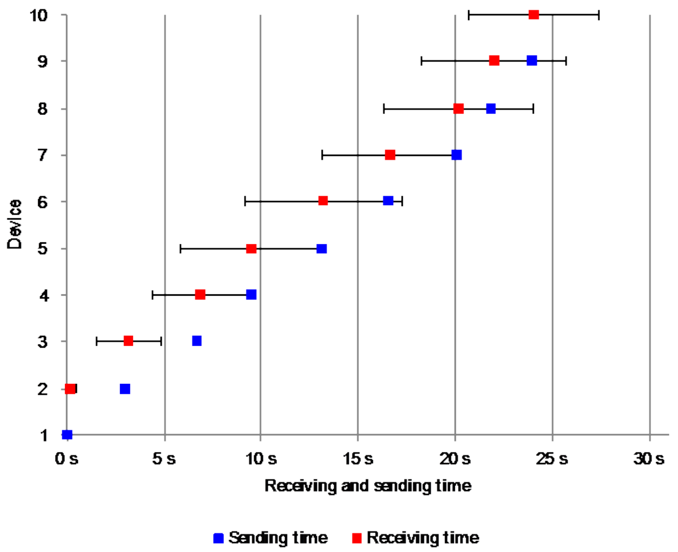

Figure 5 Hop to hop propagation of text messages. The results were averaged over 21 runs. Error bars show the standard deviation.

Deviations from this are the result of the variable time a smartphone had to wait until it received a meta data packet that triggered the text message exchange. This lead to the growth of the standard deviation with further propagation because the uncertainty sums up at every device.

4.3 Map Application

The map application covers another useful area for the priority based data dissemination. It allows the exchange of map segments between devices, while beginning with the segments that have little information density and cover a large area. In the long run highly detailed segments which contain only information about a small geographic area are added. The second parameter, despite the information density of the files, is the distance of the map segment center to the current location. Segments that are close to the current device location are transferred before remote segments.

The two criteria are merged in one continuous priority function that determines which segment has the highest priority at the moment of transmission. In the map application OpenStreetMap data is used. To achieve small file sizes and continuous zoom and to avoid redundant data, vector map data is used within the application. Therefore the OpenStreetMap XML data had to be prepared.

In order to offer segments with different information density, the main map is separated into three granularities. While granularity 0 includes only the main roads, highways, railways, ferries and rivers, granularity 1 contains smaller roads and greenland. For granularity 2 remain all other categories which are not included in the first two granularities. After the filtering every main granularity map is separated in small segments, that can be displayed individually by the rendering engine. These segments are the ones exchanged by the application between the devices.

4.3.1 Linear Propagation

a) Test Setup: The test setup of the linear propagation follows the arrangement as described in section IV-B1a. To test the map application an evaluation data set of 15 map segments with all granularities has been created.

b) Results: The on-device logging of the receiving and sending timestamps of the map segments is shown in figure 7. According to the priority based ata dissemination the map segments with high priority were sent first by the source device and arrived regularly in the same order at the last device.

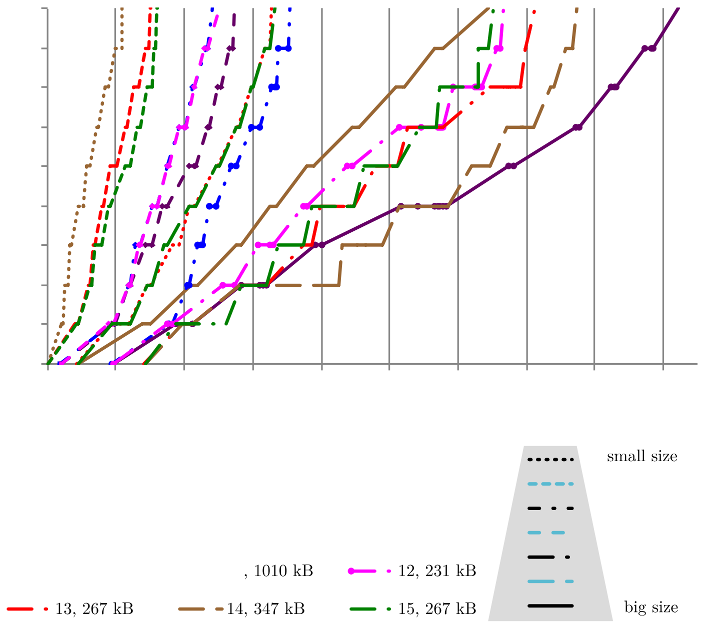

Figure 6 Hop to hop propagation of map segments. Dotted lines mark smaller segments, whereas solid lines mark bigger map segments. The number in front of the segment size indicates the priority order.

It can bee seen, that the segment size is also a factor. As every segment exceeds the MTU it is fragmented into smaller packets. The more fragments a segment has the higher is the possibility for packet loss when using the unreliable user datagram protocol (UDP). A segment with a low number of fragments can pass a segment with more fragments at any node because only complete segments are transferred. This is due to missing fragments preventing the calculation of the content sensitive priority and therefore other, lower prioritized but complete segments get the chance to be transferred first. This behavior can be seen for example with the first two segments. Although the priority of the first dataset is higher, the smaller segment can be forwarded earlier by every following node.

Furthermore one can see that both big segments can influence the transmission of following smaller segments negatively when the preferred transmission of lost packets survives multiple meta data cycles. The segments 12, 13 and 15 are obviously influenced by the big segment 8, which shows up in the widening of the graph in figure 7 to the right side.

The functionality of the priority based data dissemination is demonstrated by this test. Data segments with a high priority are transmitted first. Only the size of the single data segments and the unreliable UDP transmission could change the order of arrival at the last device.

4.3.2 Route Simulation

a) Test setup: For the second scenario a route was simulated with two devices. Both smartphones were provided with fake GPS data, that simulated the synchronous moving from the start city Ulm to the destination Biberach in Germany. This tested the dynamic change of priority when the devices move, which leads to a change of the location for the priority calculation.

In this scenario one smartphone, the source device, was equipped with the full map data set of the segmented Open- StreetMap data. The second smartphone hadn't stored any map data at test start. It had a storage limit of 15 segments set. The simulation lasted 15 minutes.

On both smartphones the simulation of the GPS signal was started simultaneously. The following start of the map application initialized the metadata transmission. This lead to the sending of the highest prioritized map segments by the source smartphone. The second smartphone stored the segments until it reached its storage limit.

The simulated movement of the two devices resulted in a dynamic modification of the segment priorities. Therefore the source device transmitted new map segments that now have the highest priorities. These segments replaced the segments on the second smartphone that were now more remote and lost their high priority.

b) Results: Figure 8a illustrates the situation shortly after the test started. The map segments around the start city Ulm and remote highways have been transmitted to the before empty device with the limited storage capacity. Please note that the map data eastern to Ulm belong to another state and are not part of the map data set.

At the end of the simulation, which is shown in figure 8b, the detailed map segments of the surrounding of Ulm have been replaced by the ones around the destination city Biberach because the latter then had a higher priority. The segments with highways of granularity 0 around Ulm still had a high priority and therefore remained on the device.

The dynamic transmission of map segments is presented in figure 9 where the map segment centers are drawn on a north-up map. The chronological sequence of the simulation is assigned by the color of the markers which is defined by a heat map from beginning to end. The shape of the markers points to the granularity of the segments. One can see that in the vicinity of the starting point Ulm red colors with early chronological assignment dominate, whereas at the end of the route green colors marked with the chronological maximum prevail. A closer look reveals that the remote segments of granularity 0 are transmitted very early in the simulation.

This corresponds to the desired effect to draw a rough picture of geographically big map segments with little information before transmitting highly detailed data. In a smaller scale the segments of granularity 1 are also higher prioritized than segments of granularity 2 and therefore transferred before the latter.

5 Conclusion and future work

In this paper we presented an approach to priority based data dissemination for mobile adhoc networks. A generic protocol, that employs a content sensitive prioritization function was proposed. The protocol consists of two parts. One part cyclically broadcasts the available information that a device currently stores. The other part is used to transmit missing data fragments via broadcast to neighboring devices.

Based on the priority function it is ensured, that higher prioritized data is distributed faster throughout the network. On the other hand a dynamic function allows data to age and ultimately be discarded in the network. If the data available to the device exceeds the device storage, the priority function provides the means to discard, or to replace packets in the storage queue.

Two applications were built on top of the protocol. One distributes short text messages throughout the network. The other application exchanges OpenStreetMap data to display a vector based map on the device. The map data was divided into different segments containing different levels of detail with varying section sizes. Utilizing a location based priority function, map segments closer to the device are exchanged first.

The protocol and the two applications were implemented as a smart phone application. Using current of the shelf hardware, the applications were tested in real world scenarios. The results show, that the priority based data dissemination is works well for both applications. On a memory constraint device the protocol in conjunction with the map application performed quite well and was able to keep the important (currently required for displaying a local map) data on the device while discarding unneeded data.

As future work we plan to build further applications based on the priority based data dissemination protocol. Also the existing applications can be improved. Especially for the text application an encryption scheme comes to mind, where public keys are distributed throughout the network and the text messages are encrypted. The protocol itself requires further improvement, as of right now there is no mechanism to prevent neighboring devices from sending non required data over and over again.

From an evaluation standpoint a further investigation into the impact of the different parameters of the protocol should be conducted. Also the limitations regarding device density, contact times or device speed should be investigated. On the other hand the benefits in terms of reduced data transmissions, energy and channel utilization should be subject to further tests.

REFERENCES

[7]. L. Pelusi, A. Passarella, and M. Conti, “Opportunistic networking: data forwarding in disconnected mobile ad hoc networks,” Communications Magazine, IEEE, vol. 44, no. 11, pp. 134–141, November 2006.

[8]. J. Ott, “404 not found: A quest for dtn applications,” in Proceedings of the Third ACM International Workshop on Mobile Opportunistic Networks, ser. MobiOpp ’12. New York, NY, USA: ACM, 2012, pp. 3–4. [Online]. Available: http://doi.acm.org/10.1145/2159576.2159579

[9]. S. Batabyal and P. Bhaumik, “A fundamental traffic management strategy for mobile opportunistic network,” in Advances in Computing, Communications and Informatics (ICACCI), 2013 International Conference on, Aug 2013, pp. 822–827.

[10]. J. Ott, E. Hyytia, P. Lassila, T. Vaegs, and J. Kangasharju, “Floating content: Information sharing in urban areas,” in Pervasive Computing and Communications (PerCom), 2011 IEEE International Conference on, March 2011, pp. 136–146.

[11]. V. Soares, F. Farahmand, and J. Rodrigues, “Scheduling and drop policies for traffic differentiation on vehicular delay-tolerant networks,” in Software, Telecommunications Computer Networks, 2009. SoftCOM 2009. 17th International Conference on, Sept 2009, pp. 353–357.

[12]. N. Muslim and F. Pregizer, “Designing a testbed for broadcast opportunistic wireless data dissemination protocols,” in Advanced Information Networking and Applications Workshops (WAINA), 2013 27th International Conference on, March 2013, pp. 607–612.

[13]. F. Pregizer and H. Klee, “Real-time audio transmission in mobile adhoc networks,” in 2013 IEEE 5th International Symposium on Wireless Vehicular Communications (WiVeC 2013), June 2013.

