The International Congress for global Science and Technology

CNIR

ICGST International Journal on Computer Network and Internet Research (CNIR)

Volume (8), Issue (II)
December, 2008,

www.icgst.com
www.icgst-amc.com
www.icgst-ees.com

AIML-CNIR Journal
ISSN Print 1687-4846
ISSN Online 1687-4854
ISSN CD-ROM 1687-4862

CNIR

Table of Contents

CNIR

ICGST International Journal on Computer Network and Internet Research (CNIR)

A publication of the International Congress for global Science and Technology (ICGST)

ICGST Editor in Chief: Dr. rer. nat. Ashraf Aboshosha

www.icgst.com, www.icgst-amc.com, www.icgst-ees.com

editor@icgst.com

www.icgst.com

New Routing Technique to improve Transmission Speed of Data Packets in Point to Point Networks

S.Anuradha *, G.Raghuram, K.E.Sreenivasa murthy, B.Gurunath Reddy
*G.Pulla Reddy Engineering College, Sri Krishnadevaraya University
Kurnool 518002 AndhraPradesh India
anuradha_sanike@yahoo.co.in
http://www.gprec.ac.in

Abstract

A number of routing algorithms based on the ant-colony optimization have been proposed for communication networks. However, there has been little information on the routing technique used in analysis of ant-routing algorithms. In this paper, a new routing technique named DB routing is discovered and we have compared the performance of DB routing algorithm an Ant-Colony Optimization, with Dijkstra's shortest path algorithm as benchmark. Our simulation results show that the performance of DB routing is comparable to the benchmark algorithm. Moreover, under varying traffic loads, DB Routing adapts to the changing traffic and performs better than shortest path routing.

Keywords: *Ant-colony optimization (ACO), DB-routing algorithm, Positive feedback, Negative feed back.*

1. Introduction

A wide range of routing algorithms and protocols exist for communication networks. In the traditional routing approach, the routing tables are updated by exchanging routing information between the routers. Various routing protocols differ in their approaches to exchange the routing information. For example, in Open Shortest Path First (OSPF) the routers exchange link-state information by flooding. A relatively new approach to routing is the use of mobile agents for updating and maintaining the routing tables. Recently, a number of routing algorithms inspired by the ant colony metaphor and mobile agents have been proposed for both wired and wireless networks. In this paper, we refer to the algorithm as DB routing algorithm (DBrouting). In general any routing algorithm will follow either depth first search technique (DFS) or breadth first search (BFS) techniques but in the DB routing the algorithm will adopt both the techniques to get the shortest path. An DB Routing Algorithm is a hop-by-hop routing algorithm based on the principle of stigmergy observed in real life [1-3] ant colonies. Stigmergy is a form of indirect communication mediated by modifications of the environment [3-14]. It has been shown that real ants are able to find the shortest path by following the trail of a chemical substance called pheromone deposited by other ants [18, 19, and 23]. The idea behind DB Routing is to use a form of stigmergy to coordinate societies of artificial agents. The artificial agents (mobile agents) or ants move on the network and are used to update the routing tables. The mobile agents update the routing tables in an asynchronous manner independent of other mobile agents. Thus, in DB Routing the routers (nodes) do not need to directly exchange routing information for updating the routing tables [15-17]. The principles of ant colony and stigmergy have been applied to numerous other optimization problems besides routing and have been referred to as ant colony optimization in the literature [6, 20, and 25, 26, 28]. There are two critical components that determine the performance of a DB routing. First, the performance depends on how the mobile agents search for the shortest path, which is referred to as exploration [16]. The mobile agents could either use heuristics based on the routing tables or the routing table values without any modifications to move to the next node. Second, the mobile agents have to update the routing tables based on the paths that have been searched. In general, exploration and routing table update in DB Routing are coupled since the mobile agents use the same routing table values for exploration which they also update. In all the DB Routing proposed so far, the algorithm parameters have been chosen heuristically. In this paper, we analyze the the routing technique and we compare the performance of DB Routing with Dijkstra's shortest path algorithm for constant delays on the performance of Ant Net algorithm. Our simulations indicate that the performance of the DB Routing algorithm is comparable to the Dijkstra's shortest path algorithm under non-varying traffic. However, the performance of the DB Routing algorithm is dependant on the network size and topology. One of the key insights is that the DB Routing algorithm performs well for grid topologies.

The rest of this paper is organized as follows. Section 2 discusses the Ant colony optimization. Section 2.1 Presents Routing strategy for proposed work. Section 2.2 Presents Proposed routing algorithm pseudo code. Section 2.3 presents Network model used in DB routing. Section 2.4 Presents Network topology used in DB routing. Section 2.5 Presents Complexity of DB routing. Section 2.6 Presents results and Discussion. Section 3 Presents Conclusion of the work.

2. Ant Colony Optimization, an Overview

Ants are able to find the best way to and from food source, in spite of being practically blind. Ants deposit a chemical substance called pheromone for marking path from food source to nest. By sensing pheromone trails, foragers can follow the path to food discovered by other ants. Shortest routes have high concentration of pheromone and almost all ants end up using this path. This behavior has been confirmed by Double bridge experiment by Deneubourg et al [2]. This behavior demonstrates collective and unsupervised learning without any centralized control. This simple but effective collective trail-laying and trail-following behavior is the inspiring source of ACO. The idea behind ant algorithms is to simulate artificial stigmergy to coordinate societies of artificial ants. Initially, the ants choose one of the available paths randomly because of absence of pheromone trail on any of the paths. However, the ants that choose shorter paths, take less time to traverse it and hence pheromone deposition on the shorter paths occur earlier than the longer ones. Ants which arrive after pheromone deposition on shorter path has occurred and before ants on longer path have completed their journey prefer to choose shorter path because of higher pheromone concentration on it. More number of ants on shorter path further increases the rate of pheromone deposition on that path. Cumulatively, over a period of time this results in highest pheromone concentration on best path and finally all the ants travel through that path.
The Ant Colony Optimization (ACO) metaheuristic is a recently proposed discrete optimization metaheuristic for solving NP-hard problems. In the following we will present a formal definition of the problem representation on which ACO algorithms work. We will describe formally how solutions are constructed in the representation and present the ACO metaheuristic as pseudo-code. The central properties of ACO are based upon the self-organized collective foraging behavior of ants. The key property collective in the foraging behavior of ants is their ability to find shortest paths between the location of their anthill and the location of food sources. When the ants move on a path between their anthill and the location of a food source they lay a pheromone trail. Other ants can then follow these generated paths of pheromone trails. This means that ants tend to converge on the same path, In ACO; solutions are constructed repetitively by adding solution components to partial solutions stochastically. Solutions are constructed by taking into account (i) heuristic information when adding solution components (if available), and (ii) (artificial) pheromone trails which change dynamically based on the experience of the ants. Stigmergy handles the propagation of experience between ants. As related in the brief historical survey above, a number of algorithms following the ACO metaheuristic have been presented in recent years. The problems solved can be divided into two domains – static and dynamic problems. The basic properties of the algorithm do not change because of this, and the problems are in both cases represented by graphs. The ACO metaheuristic has been used as a template for algorithms that have achieved world-class performance6 in both domains. Nevertheless, because of the ants' inherent ability to adapt to changes in the environment, we find that ACO algorithms are especially well suited to solving for example routing problems in networks, which constitute a dynamic problem domain.
The main reason for choosing the Ant Colony Optimization as the base for DB routing is, in general any routing algorithm will follow either Depth first search (DFS) or Breadth first search (BFS) technique for routing the packets in the Networks. Each of the techniques have its own merits and demerits but in the Ant system of Ant Colony Optimization ants will spread throughout the Network for searching the better paths and moreover it takes the DFS and BFS methods into consideration and finds the better paths which were not highlighted in the original Ant system. Some modifications were made to suit for adopting the combination of the two searching techniques in the proposed algorithm. Hence it is named as Depth wise Breadth wise searching algorithm (DB routing algorithm.)

2.1 Routing Strategy of the Proposed Work

Many strategies have been proposed and researched for Network routing. For the purposes of this paper, two Characteristics of such strategies are considered important. Firstly, networks are either packet or circuit switched. A packet switched network is one in which routing decisions are made on a packet-by-packet basis. In this case, no fixed connections are made and resources are not reserved. In a circuit switched network a connection is maintained for the duration of a session between two or more entities. Resources are allocated to the circuit for the duration of the session. Secondly, network routing can be either static or adaptive. Static routing usually employs shortest path algorithms such as provided by Dijkstra's algorithm in order to compute routing tables that are subsequently downloaded to the network. Adaptive routing uses link cost metrics, which are the functions of the utilization of network resources in order to force changes in routing during periods of network congestion. Considerable information on routing can be found in [Schwarz 89] and [Tanenbaum 88]. The motivation for exploiting the ant metaphor for routing in telecommunications networks arises from the fact that routing systems frequently depend upon global information for their efficient operation. Ant systems do not need such global information, relying instead upon pheromone traces, or rather their digital equivalent, that are laid down in the network as the ant, or agent, moves through the network. Global information is frequently out Of date and transmission of the information required from one node to all others consumes considerable network resources. Ideally, we would like to have the network adapt routing patterns to take advantage of free resources and move traffic if possible. This is particularly desirable in broadband networks where traffic patterns change rapidly and maintaining a global view of available network resources is almost impossible. To date, three applications of the ant metaphor in the domain of routing have been documented [White 97], [Schoonderwoerd et al 97] and [Di Caro and Dorigo 97]. Schoonderwoerd's work embraces routing in the circuit switched networks while Di Caro and Dorigo deal with packet switched networks. Di Caro and Dorigo, in particular, provide compelling experimental evidence, based upon on simulation, as to the utility of ant search in the network routing problem domain by comparing ant-based routing with the current and proposed routing schemes used in NSFNET.

2.1.1 DB-Routing, an Overview

DB routing algorithm implements the mentioned strategy by using the following mechanisms:
Local heuristic value rule, transition probability rule, random step, positive feedback and negative feedback.
We will give an informal presentation of their use in DB routing, and describe how they participate in solving the Routing problem. The local heuristic value used in DB routing works by estimating the time that is taken by it to transfer a packet to an adjacent router, by utilizing knowledge about the speed of the wire and the load on the output queue connected to the wire. The estimate is used to decrease the probability of selecting wires with high use, thus reducing the risk of packet loss. This also has the effect of promoting exploration, since the wires that are highly used are, other things being equal, also the wires with the highest trail value and the longest queues.
Positive feedback is, as always in ACO algorithms, used to ensure that better paths are given a higher trail value and thus a higher probability of being used. The quality of a path is inversely proportional to the time a packet has spent being routed from the paths source to its destination. Positive feedback is in DBrouting implemented in the way that all the routers on the used path are rewarded, by an increase in trail value according to the quality of the used path. The stochastic state transition rule in DB uses local heuristics and trail value to create paths. We use two parameters α and β to determine the relative influence of local heuristic and trail value. Uniformly random steps are used in DB routing to counteract stagnation. At a given router, there exists a small probability, given as an argument to the algorithm that the packet is routed completely at random (but not to any router it has already visited).
Negative feedback in DB routing is implemented by decrementing trail values where packets are lost. This leads traffic away from bad paths, and thus lessens the risk of packet loss. This mechanism counteracts convergence and subsidiary promotes exploration since it lowers the probability of selecting a used wire thus increasing the probability of selecting the other wires leading away from the router.
All these mechanisms participate in promoting that packets are routed over good paths and in reducing the Risk of them becomes lost. Positive feedback continuously enforces the trail values in the routing table to promote the use of the best known paths. Negative feedback are used to counteract the possibility that positive feedback makes the system stagnate, and also makes it possible for the system to react if a good path suddenly becomes bad or overloaded. Negative feedback caused by packet loss and long output queues are also used to promote exploration of new paths. A number of parameters determine the importance of the different mechanisms in DB routing

2.1.2 Network Requirements

We have modeled and implemented a simulated network This simulated network is the domain in which the implemented routing algorithm
By modeling a network and letting the algorithm work with representations of objects in a physical network, we were forced to think in terms of what is possible in a "real" network. The algorithm that we have implemented requires a few functionalities in the network protocol that are not commonly in use.

• *Routing tables:* To support the stochastic path-selection performed in ACO-algorithms, it is necessary to increase the amount of information stored in a routing table. The general problem of routing from any router $s \in R$ to any router $d \in R$ $\{s\}$ in a network $N = (R, W)$ can be seen as $|R|$ problem instances in the same network. In deterministic routing algorithms, it suffices to point to the next router on the path to d – which effectively translates to a routing table containing $|R|-1$ entries (no need to route to oneself). In stochastic routing, it is necessary to support the path selection algorithm by storing information in the routing table regarding the quality of selecting each outgoing wire. The amount of space required in routing table increases, as the network-graph becomes increasingly dense. This means that in a fully connected network the routing table at each router will contain $(R-1) \times (R-1)$ entries. The routing tables implemented for our algorithm are organized in this way and contain entries that hold a trail-value that is used in the stochastic selection process.

• *Ants are packets:* All the packets, which are used to route in DB routing, are ants. As every ant maintains a memory that is used as a taboo-list, and for retracing the path to update trail-values, this memory has to be stored in the packet – either as payload or as a designated header-field.

• *Instant router update:* We use a delayed update of trail-values, so information regarding the quality of an ant's path should somehow be propagated back through the network to the routers on the used path. In our implementation, this propagation is not effectuated, only simulated. Instead, all routers are instantly informed of the quality of the path by the metaheuristic.

There are three ant agent types in the DB Routing by Ants. These are explorer ants, allocator ants and Deallocator ants. Each agent type possesses a small memory to store the route being traversed and the constraints on routing. Explorer ant agents exhibit the foraging behavior of ants and preferentially follow trails of pheromones laid down by previous explorers. Constraints on routing allow for the inclusion or exclusion of specific nodes on the route, or the exclusion or inclusion of certain links. The Allocators traverse the path determined by explorers and allocate network resources on the nodes and links used in the path. Similarly, when the path is no longer required, deallocators traverse the path and deallocate the network resources used on the nodes and links. The system works in the following way. A connection request is generated at a given node, the source. The connection request is either a point to point (P2P) For P2P requests, a new species of ant (agent) is created or m is sent out into the network. These explorer agents execute the following pseudo-code algorithm

2.2 The basic DB Algorithm

```
 1. Initialize the route finding method
     Set t: = 0
     For every edge (i, j) set an initial value Tijk (t)
     Of zero for trail intensity. Place m ants on the source
     node. {Create new explorers agents at a frequency ef }
 2. Set s:= 1 { taboo list index }
    for k:= 1 to m do
    Place starting node of the kth ant in Tabook[s].
 3. Repeat until destination reached:
    Set s := s + 1
    For k:=1 to m do
    Choose node j to move to with probability pij
     k (t)
    Move the kth ant to node j.
    Update explorer route cost: rk = rk + C(i,j)
    If (rk > rmax) then
    Kill explorerk
     Insert node j in Tabook[s].
    At destination go to 4.
 4. While s > 1
    Traverse edge Tabook[s].
    Tijk(t) = Tijk(t) + phk
    s := s - 1
 5. At source node do:
   If the path in Tabook is the same as p% of paths in
   PathBuffer then Create and send an allocator agent
   If t > Tmax then
  Create and send an allocator agent
```

In the above algorithm, the following symbols are used

- Tij(t) is the quantity of pheromone present on the
  link between the ith and jth nodes,
- C(i,j) is the cost associated with the link between the
  ith and jth nodes.
- rk is the cost of the route for the kth explorer agent.
- Tabook is the list of edges traversed.
- Tmax is the maximum time that is allowed for a Path to emerge.
- Path Buffer is the array of paths obtained by the (up to m) explorer agents.
- rmax is the maximum allowed cost of a route.
- phk is the quantity of pheromone laid by the kth explorer agent.
- pij k (t) is the probability that the kth agent will choose the edge from the ith to the jth node as its next hop.

The probability with which an explorer agent (k) chooses a node j to move to when currently at the ith node at time t is given by:

pij k (t) = [Tijk(t)]a[C(i,j)]-ß / Nk

Nk = Sj in (S(i)- Tabook) [Tijk(t)]a[C(i,j)] -ß

where a and b are control constants and determine the Sensitivity of the search to pheromone concentration and link cost respectively. Nk is simply a normalization factor that makes pij k (t) a true probability. S(i) is the set of integers, {l} such that there exists a link between the ith and lth nodes.

The central parts of implementing a DB routing algorithm is as given above, specifying a stochastic state transition rule and a pheromone trail update rule together with a trail deposition rule. In DB, ants will start at a randomly determined vertex and continue to move to another vertex until they have completed a tour. Further, the ants contain a 'memory', which is used to implement a taboo-list so that only feasible solutions are generated . This also fits well with the fact that the name DB now denotes the version, where the pheromone is deposited after a complete tour. In DB the depositing of the pheromone is proportional to the inverse of the length of the ants' tour. This has the effect that shorter tours are rewarded with larger amounts of trail.

2.3 Network Model applied in DB Routing

A rough sketch of a router in our network model is included in Figure 1.

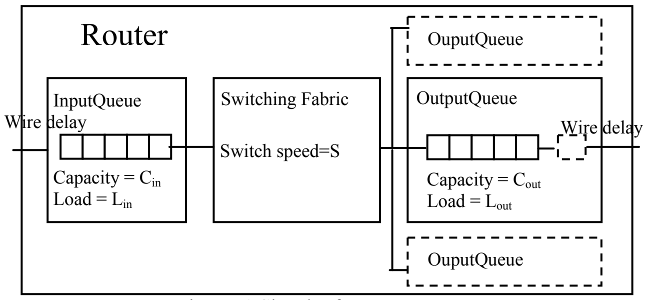

Figure. 1 Sketch of a router

A router contains an input queue with a max-capacity C_{in}, a switching fabric with a speed S, a number of output queues with a capacity C_{out} and for every output queue a transmission buffer that is used by a packet waiting to be sent over a wire. The functions L_{in} and L_{out} return at any time t the number of packets in input queues and output queues respectively. Any number of wires can be connected to a router. A relevant detail outlined in Fig.1 is that routers use separate output queues for each outgoing wire, but all incoming wires share the same input queue. Finally, we define the start and end-point for the routing of a packet to be the switching fabric of a router, this means that the packet routing is initialized by being processed by a switching fabric at the source router and that it ends in the switching fabric at the destination router.Packet loss happens in the situation where a packet could not reach its intended destination. This happens if a packet arrives at a queue, which is filled to capacity. A router has two types of queues, the input queue and the output queues. It is obvious that the input queue can be filled, thus resulting in packet loss, but one could argue that if an output queue is full, the router should wait until the queue could accept a packet. This situation is called head of line blocking, being the situation where the first element in an input queue, after being through the switching fabric cannot be inserted into an output queue. This means that in a more accurate model, either the packet should be moved to an extra buffer or the router simply waits until the needed output queue is available. To avoid head of line blocking, we assume that packets are extracted from the input queue, processed by the switching fabric and inserted into an output queue. The packet is lost if the insertion of the packet exceeds the capacity of the output queue. Another source of packet loss is bad routing. Potentially a packet can be routed indefinitely by a cycle in the routing tables, resulting in the packet never reaching its destination, and increased load on the network. A method often used to avoid this problem is a Time to Live (TTL) counter, which is decremented every time a packet arrives at a router. If the counter reaches zero, the packet is discarded and assumed lost. Packets in the network model also contain a TTL counter to avoid the problem of indefinite routing. The initial value of the TTL counter differs from network to network, but in general it should be based on the amount of routers that packets in general are assumed to visit. The job of a good routing algorithm is to reduce the amount of lost packets resulting from queues being filled and avoid that TTL counters reach zero. Another often used parameter for routing quality is the throughput in the network. Throughput describes the amount of data that can be transferred in the network as a whole. A network has a high throughput if packets are routed fast through the network and few of the packets are lost. The time spent being routed is determined by the speed of the used routers and the load on the routers. The goal of a routing algorithm is therefore to maximize throughput by reducing the average amount of time it takes to route a packet in the network while avoiding packet loss resulting from overloading single routers. It should be evident from the above definition(s) and discussions that a network represents a dynamic problem domain as the lengths of the paths fluctuate over time. We feel it important to stress the subtleties of the dynamics in the network since they differ somewhat from the dynamics found, had we simply used a dynamic graph with edge weights being stochastic functions only of time. The relevant observation to make is that the conditions for movement are different. In a dynamic graph as defined in the previous paragraph, the ants must also react to changes in the domain in which they work. However, the work they perform does not affect the underlying domain as it only changes as a function of time. In a network, this is not the case. One could state that delays change deterministically. Whenever a path between two routers is constructed, and packets are sent over this path, the length of this path changes as a function of how many packets are sent to routers on this path. The situation becomes even more complex by the fact that a network is highly distributed consisting of a number of routers working concurrently, which in collaboration must find the best way to route all packets Sent. Each time a router sends a packet to a given target router, this target router's input queue load is incremented, which in essence effects all other routers in the network since their path length to that router has been increased by the increase in input queue length on the previously mentioned target router. The described 'feedback mechanism', does not exist in the dynamic graph mentioned above, where limitations in capacity and delays do not exist. This essentially sums up the main reason, why we decided to design a fully working simulated network. We could have slightly extended our initial implementations, based on our definitions of a graph to vary the edge-costs dynamically, thereby creating a very limited simulation of a network. An implementation of this sort would not have had the feedback mechanism described above, and furthermore this would not have made us able to define the quality parameters as easily.

2.4 Network Topology applied in DB-Routing

The basic performance is tested and abilities of DB routing upon a network topology derived from the double bridge experiment. The network consists of four routers connected by four wires as shown in Figure.2.

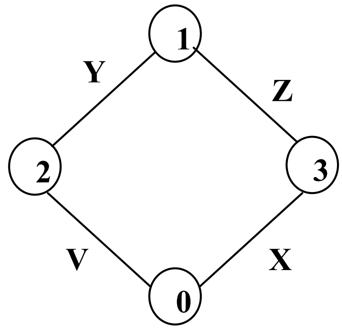

Figure. 2 The double bridge network

The router 0 is the sender of packets in the simulations with this network topology. The input queue of 0 is set to an arbitrarily high value so that – no matter the creation interval of packets – packets will not be rejected from the network. In effect, this is a buffer for recently created packets. This means that the packet-switching delay of this router decide the rate at which packets flow into the rest of the network (this delay is set to 7). The routers 2 and 3 have set input queue and output queue-sizes with space enough for 10 packets. The packet-switching delay of these routers is set to 10. The router 1 is the receiving router in simulations with this network topology. We have conducted experiments with an input queue length of 10 packets (as above), but the packet-switching delay is only 1, so that the router does not drop packets from its input queue (as a packet is removed from its input queue at every time step). Furthermore experiments were performed on a 10x10 grid network, which is shown in Figure.3. It consists of 100 routers connected in a grid with wires (with a delay of 10) between adjacent routers. We used this topology to test DB ability to route in a 'larger' (but still simple) network.

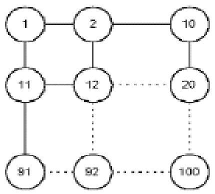

Figure. 3: The 10x10 grid network

The router 23 is the sender of packets in the simulations with this network topology. Router 23 is similar to router 0 in the double bridge network. The router 85 is the receiving router in simulations with this network topology. Again this router is similar to the receiving router above. The rest of the routers have set input queue and output queue-sizes with space enough for 10 packets. The packet-switching delay of these routers is set to 15. The reason why we have chosen these two routers is simply that they are internal routers in the network Topology – making as many feasible paths as possible. The exact location of the routers was somewhat arbitrarily chosen. This topology furthermore has the property that a number of paths are of equal initial quality. All paths in the rectangle spanned by router 23 and 85, which only consists of moves to a router with a higher index (moves to the "right" and "down"), will initially be of same length.

2.5 The Complexity of DB-Routing

Now we calculate the complexity of DB routing:

M represents number of ants, Nk represents the set of neighbors of the current node k

We first calculate the complexity, defined as number of elementary operations, of a single ant to travel between a given source node and a given destination node in the AntNet algorithm. At every node along the path between a given source and destination node, the ant needs to search through the stack it maintains in the memory to find whether to use for choosing the next node or to choose the next node uniformly among the neighbors. The worst-case complexity of searching through the stack is O (1), if the stack is implemented as a combination of linked list and an additional array or a hash table. Further, the complexity is O (Nk) since the probability values have to be calculated for each of the Nk neighbors. In DB routing, the ants choose the next node uniformly among the neighbors of the node, and therefore the above operations are not required. There are other computations that an ant performs at each node: the ant needs to push the identifier of the current node and the time at which it arrived into the stack which is O(1), the ant goes to the queue for one of the outgoing links which is O(1). Let the maximum hop count of the ant be M. In the worst case, the ant has to do all the computations at each of the M nodes. Thus, the worst-case complexity for a single ant to travel between a given source node and a destination node in Ant Net is O (MNk),

While in DB routing it is O (M). The worst-case complexity for a single ant to travel between a given source node and a destination node is also O (MNk). This can be calculated as follows.

The ant performs three operations at each node along the path it travels between a given source node and a given destination node. First, the ant needs to pop the stack to find out the next node to travel. The pop operation in the stack is O (1). Second, the ant goes to the queue for one of the outgoing links which is O (1). The ant updates the routing table for each of the neighbors Nk for the destination d which is O (Nk). Furthermore, in the worst case the ant has to do all the computations at each of the M nodes. We calculate the worst-case complexity of DB routing when the total number of ants generated is given by $q > 1$. Since the worst-case complexity of a single ant to travel between a given pair of nodes is O (MNk), furthermore, we know that the worst-case complexity for Dijkstra's shortest path algorithm using a Fibonacci heap is O (NlogN + L).

This analysis shows that the complexity of DB routing algorithm is less than original ant system and is comparable to Dijkstra's' shortest path algorithm when M and Nk are small as compared to N.

2.6 Experimental Results and Discussion

2.6.1 Results of Experiment 1 – testing of basic Capabilities in a 10x10 Lattice Network

The intension was to show the basic abilities of DB routing by altering some of the parameters. In experiment 1.1 the trail was prioritized above local heuristics, we therefore expected the path of the packets to converge more or less to a *single path* and that the packets would die on the queues of the routers. As can be seen on Figure 4 the packets mainly follows two paths (both short paths), so our assumption was not completely wrong. As expected almost all of the dead packets died on the input queues of the routers.

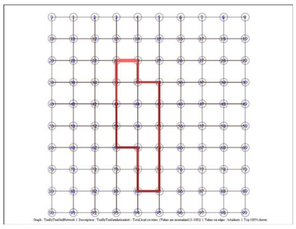

Figure 4: The paths generated from experiment 1.1

Number of packets	10000
Dead	3430
TTL	0
Input	3050
Output	215
NPR Dies	164
Average time for Packet	453.78

Table 1: Results of Experiment 1.1

In experiment 1.2 we reversed the settings, so we gave local heuristics a higher priority than trail. Because of this setting we predicted that the packets are distributed over the entire network and that they had a larger probability of dying of no-possible-routing than in experiment 1.1. As anticipated the packets have no favorite path and most of the network is traversed by packets (see Figure 5). The distribution of where the packets are lost is also shifted, so that approximately an even number is lost on input queues and due to no-possible-routing.

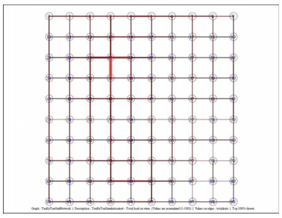

Figure.5: The paths generated from experiment 1.2

Number of packets	5000
Dead	2866
TTL	0
Input	1285
Output	0
NPR Dies	1580
Average time for Packet	675.9

Table 2: Results of Experiment 1.2

In the experiment 1.3 (see Figure 6) we basically performed experiment 1.1 again just with negative feedback, so that the algorithm should be able to react to loss of a packet. We predicted that the packets were routed by a few good paths, but that these would be sidestepped for a time-period when packets were lost. The negative feedback did as expected and forced the packets to follow more paths than in experiment 1.1. And the paths are moreover longer. The negative feedback also reduced the number of lost packets as expected (to 11% of the original number).

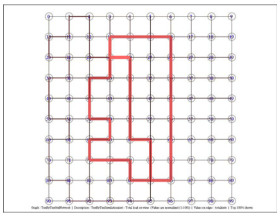

Figure.6: The paths generated from experiment 1.3

Number of packets	5000
Dead	362
TTL	0
Input	216
Output	0
NPR Dies	148
Average time for Packet	400.12

Table 3: Results of Experiment 1.3

Experiment 1.4 is experiment 1.2 with negative feedback. We expected that this would force the packets to be routed to routers with a low load. As can be seen on Figure 7 the packets are distributed over the entire network, as a result from this almost 3/5 of the packets were lost in the network primarily due to no possible-routing.

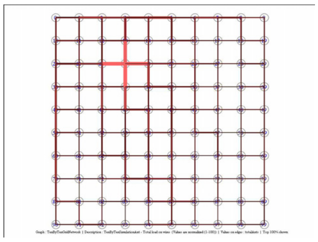

Figure.7: The paths generated from experiment 1.4

Number of packets	5000
Dead	2849
TTL	0
Input	851
Output	0
NPR Dies	1999
Average time for Packet	790.38

Table 4: Results of Experiment 1.4

Analysis of the First Experiment

We have shown that it is possible to control how DB routing routes in a (grid) network by varying the parameters given to DB routing. By varying α and β relatively to each other it is possible to control if DB routing should send all the traffic by some highly loaded paths or if it should distribute the load out on more routers by not converging to just some paths. By also adjusting on negative feedback it is possible to reduce the loss of packets considerably, but this also forces DB routing to utilize more routers/paths. This also increases the average time that a packet spends in the network.

The adjustments to the different parameters also greatly affect which type of packet deaths we experience.

If the packets are forced along some highly loaded paths, the packets are lost on the queues of the routers.

The more the packets are distributed the greater is the possibility of no-possible-routing deaths, because of the packets reaching a dead end due to their taboo-lists. There is no possibility of packets dying because of TTL in DB routing. This is because of the taboo-list and the size of TTL, which allows the packets only to visit *x* routers, where *x* is equal to the number of routers in the network. So TTL could only happen if a packet visited a router twice (this is not allowed by the taboo-list).

2.6.2 Experiment 2 – Double Bridge Network with heavy Network Load

In experiment 2.1, where all wires had an equal delay, we expected to find that the system did not converge to a single path, because a single path could not handle the entire load in the network. As can be seen on Figure 8 (because the experiments 2.1, 2.2, and 2.3 all generated approximately the same picture, we decided to use only one picture for all three experiments) most of the traffic is handled by the left branch and the remaining traffic is handled by the right branch. In experiment 2.2 the picture does not change much. In this experiment we have a larger delay on wire *X* and this forces more traffic to the left branch and the remaining traffic is handled by the right branch, which we expected. The result of this can be seen on Figure 8. As we stated under our theses for experiments 2.3, we predicted that the outcome of experiment 2.3 should be the same as the outcome of experiment 2.2, which showed to be correct.

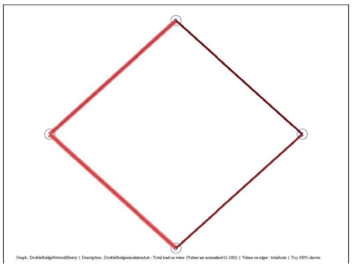

Figure.8: The paths generated from experiment 2.1, 2.2, 2.3

2.1	
Number of packets	1000
Dead	33
TTL	0
Input	0
Output	33
NPR Dies	0
Average time for Packet	1075.66

2.2	
Number of packets	1000
Dead	148
TTL	0
Input	0
Output	148
NPR Dies	0
Average time for Packet	1163.69

2.3	
Number of packets	1000
Dead	152
TTL	0
Input	0
Output	152
NPR Dies	0
Average time for Packet	1080.66

Table 5: Results from experiment 2.1 & 2.2 & 2.3

The Figure 9 shows the result of running our 'benchmark' algorithm on the setup of experiment 2.4, where the wires *X* and *Y* has a longer delay than *V* and *Z*. We allowed the 'benchmark' algorithm to update all its routing tables every 50 time-step in the simulation. We predicted that to fully utilize the network, the routing algorithm had to use both branches – and this is done by our 'benchmark' algorithm. Our 'benchmark' algorithm handles the job of routing the packets such that it 'only' losses 19% of them, which is near-optimal due to the load of the network.

Figure 10 from our experiment 2.4 where DB is run shows a result similar to Figure 9. Both the branches are utilized in order to fully exploit the network. The primary branch is this time the left branch (the opposite of the 'benchmark' algorithm) because of the local heuristic. DB almost handles the job of routing as well as the 'benchmark' algorithm as it 'only' losses 24% of the packets.

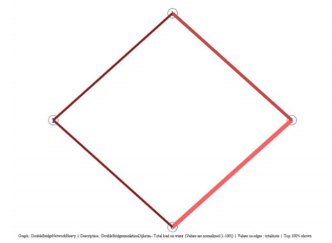

Figure 9: The paths generated by our 'benchmark' algorithm from experiment 2.4

Number of packets	1000
Dead	186
TTL	0
Input	0
Output	186
NPR Dies	0
Average time for Packet	1135.09

Table 6: Results from experiment 2.4 'bench mark'

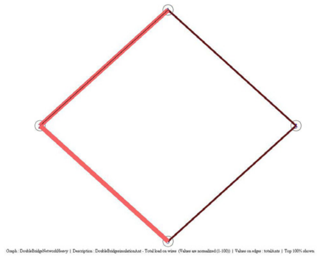

Figure.10: The paths generated from experiment 2.4.

Number of packets	1000
Dead	239
TTL	0
Input	0
Output	239
NPR Dies	0
Average time for Packet	1151.09

Table 7: Results from experiment 2.4 "DB routing"

In this experiment we wanted to show that in a heavily loaded simple network DB routing is able to utilize several paths to achieve a better throughput than by just using a single path. All our tests have shown that DB routing is able to route most of the traffic by one branch (until it cannot handle more) and the remaining traffic by the other branch. Compared to our 'benchmark'-algorithm, which was able to only loose 19% of the packets, DB routing performs a little worse by loosing 24%, but because of the network setup this is quite reasonable.

2.6.3 Experiment 3 –10x10 Lattice Network with heavy Network Load

Our third experiment was carried out in the 10x10-grid network where the network load was heavy. In this Experiment we ran both DB routing and our 'benchmark'-algorithm and compared the results. Like previously we allowed our 'benchmark'-algorithm to update all its routing tables at every 50th time-step. The Figure 11 shows how our 'benchmark'-algorithm chose to route the packets. As can be seen on the picture the algorithm distributes the packets over most of the routers in the 'center' of the network and there are no clear routes except near the start and destination routers.

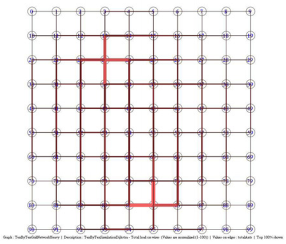

Figure.11: The paths generated from experiment 3 by the 'bench mark algorithm'

Number of packets	3000
Dead	14
TTL	0
Input	11
Output	3
NPR Dies	0
Average time for Packet	573.62

Table 8: Results from experiment 3 'Bench mark'

DB routing Figure 12 on the other hand chooses to utilize two short paths and two longer paths to route the Packets by. DB routing also routes more traffic by the two short paths and less traffic by the two longer paths (this can be difficult to **be** seen on the pictures when they are not in color).

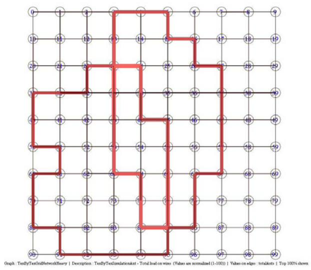

Figure.12: The paths generated from experiment 3 by 'DB routing'

Number of packets	3000
Dead	383
TTL	0
Input	175
Output	4
NPR Dies	204
Average time for Packet	540.61

Table 9: Results from experiment 3 'DB routing'

2.6.4 Comparison of DB Routing and the Benchmark Algorithm

As can be seen in Table 10 of the 3000 packets sent from router 23 to router 85 our 'benchmark' algorithm Only drops 14 of the packets (approximately 0.5%), while DB routing drops 383 packets (approximately 12.7%). A great deal of the packets dropped by DB routing (about 280), are dropped in the initial convergence The types of packet death support this, because NPRDies (death from no-possible-routing) indicates that the packets have reached a dead end due to their taboo-list.

	DB routing	Benchmark
Number of packets	3000	3000
Dead	383	14
TTL	0	0
Input	175	11
Output	4	3
NPR Dies	204	0
Average time for Packet	540.61	573.62

Table10: Comparison of DB routing and the 'benchmark' algorithm

3. Conclusion

A surprising observation is that DB is able to route the packets faster through the network than the 'benchmark' algorithm. The reason for this is the difference in the way the traffic is distributed (see Figure 11 and Figure 12). The 'benchmark' algorithm distributes the traffic by utilizing most of the routers in the middle of the network without any clear pattern and DB routing distributes the traffic by using four different paths. The reason why our benchmark algorithm is slower is that it is recalculated every 50 time-step. Because of this continuous recalculation (based partly of the load of the path) the routing tables are always changing. So it is quite possible that a packet is routed back and forth between two routers (with approximately the same delay to the destination router) due to the loads on the neighbor routers. This problem occurs because we include load into the delay calculation for the path, so that the routing tables oscillate between several paths to the destination. This could force our 'benchmark' algorithm into problems with TTL, but because TTL is in our algorithms as default equal to the number of routers in the network (100 in this case) this did not happen. This can also be seen in Table 10 where none of the algorithms losses packets due to TTL. Clearly there is a compromise between getting all the packets through the network and getting them fast through the network.

4. References

[1] Colorni, A., Dorigo M. & Maniezzo V. (1996). "The Ant System: Optimization by a colony of cooperating agents", *IEEE Transactions on Systems, Man, and Cybernetics*-part B, Vol. 26, No.1, pp.1-13.

[2] Nigel Bean, Andre Costa, An analytic modelling approach for network routing algorithms that use ''ant-like'' mobile agents, Computer Networks 49 (2005) 243–268.

[3] Daniel Camara, Antonia A.F. Loureiro, Ants A novel routing algorithm for ad hoc networks, in: Proceedings of

the 33rd Hawaii International Conference on System Sciences, 2000.

[4] Gianni Di Caro, Marco Dorigo, AntNet: distributed stigmergetic control for communication networks, Journal of Artificial Intelligence Research 9 (1998) 317–365.

[5] Andre Costa, Analytic Modelling of Agent-based Network Routing Algorithms, Ph.D. thesis, School of Applied Mathematics, University of Adelaide, 2003.

[6] Marco Dorigo, Thomas Stu¨ tzle, Ant Colony Optimization, MIT Press, Cambridge, MA, 2004.

[7] Mesut Gu¨nes_, Otto Spaniel, Routing algorithms for mobile multi-hop ad-hoc networks, network control and engineering for QoS, security and mobility II, IFIP TC6/WG6.2 & WG6.7 Conference on Network Control and Engineering for QoS (Net-Con 2003), 2003. pp. 120–138.

[8] R. van der Hofstad, G. Hooghiemstra, P. Van Mieghem, First passage percolation on the random graph, Probability in the Engineering and Informational Sciences (PIES) vol. 15 (2001) 225–237.

[9] P. Van Mieghem, Performance Analysis of Computer Systems and Networks, Cambridge University Press, 2005.

[10] Martin Roth, Stephen Wicker, Termite: emergent ad-hoc networking, in: Proceedings of the 2nd Mediterranean Workshop on Ad-Hoc Networking, June 2003.

[11] Gianni Di Caro, Marco Dorigo, Two ant colony algorithms for best-effort routing in datagram networks, in: Proceedings of the 10th International Conference on Parallel and Distributed Computing and Systems, 1998.

[12] Ruud Schoonderwoerd, Owen Holland, Janet Bruten, Antlike agents for load balancing in telecommunication networks, in: Proceedings of the First International Conference on Autonomous agents, 1997.

[13] Devika Subramanian, Peter Druschel, Johnny Chen, Ants and reinforcement learning: a case study in routing in dynamic networks, in: Proceedings of the 2nd Mediterranean Workshop on Ad-Hoc Networking, June 2003.

[14] Richard S. Sutton, Andrew G. Barto, Reinforcement Learning: An Introduction, MIT Press, Cambridge, MA, 1998.

[15] Gutjahr, W. J. (2000). "A Graph-based Ant System and its convergence", *FGCS* **16** pp. 873-888.

[16] L.P. Kaeilbling, M.L. Littman, A.W. Moore, Reinforcement learning: a survey, Journal of Artificial Intelligence Research 4 (1996) 237–285.

[17] Alberto Leon-Garcia, Probability and Random Processes for Electrical Engineering, Addison-Wesley Publishing Company, 1994, ISBN 0-201-50037-X. 1996.

[18] R. Beckers, J.L. Deneubourg, S. Goss, Trails and U -turns in the selection of the shortest paths by the ant Lasius Niger, Journal of Theoretical Biology 159 (1992) 397–415.

[19] J.-L. Deneubourg, S. Aron, S. Goss, J.-M. Pasteels, The selforganizing exploratory pattern of the argenttine ant, Journal of Insect Behavior 3 (1990) 159–168.

[20] Marco Dorigo, Gianni Di Caro, L.M. Gambardella, Ant algorithms for discrete optimization, Artificial Life 5 (2) (1999) 137–172.

[21] F. Ducatelle, Gianni Di Caro, L.M. Gambardella, An t agents for hybrid multipath routing in mobile ad hoc networks, in: Proceedings of the 2nd Annual Conference on Wireless On demand Network Systems and Services (WONS), 2005.

[22] J.M. Pasteels, J.L. Deneubourg, S. Goss, Self-organization mechanisms in ant societies (I): trail recruitment to newly discovered food sources, Experientia Supplementum 54 (1987) 155–175.

[23] S. Rajagopalan, C. Shen, ANSI: a unicast routing protocol for mobile ad hoc networks using swarm intelligence, in: Proceedings of the International Conference on Artificial Intelligence, 2005. pp. 24–27.

[24] G. Theraulaz, E. Bonabeau, A brief history of stigmergy, Artificial Life (Special Issue on Stigmergy) (1999) 97–116.

[25] Marco Dorigo, Gianni Di Caro, The Ant Colony Optimization Meta-Heuristic, McGraw-Hill Press, 1999, pp. 11–32.

[26] Cormen, T. H., Leiserson C. E., Rivest R. L. & Stein C, Introduction to Algorithms (Second Edition), MIT Press/McGraw-Hil, 2001

[27] Tannenbaum, A. S, Computer Networks (Third Edition), Prentice Hall, 1996.

[28] V.Laxmi (WCSN 2006) "Ant Colony Optimization based Routing on *ns*-2*"

Biographies

S. Anuradha obatained her B.Sc and MCA degrees from Osmania University, Hyderabad in the year 1994 and 1997 respectively.She is currently pursuing Ph.D at Sri Krishna Devaraya University, Anantapur, India. She is presently working as Associate Professor in the Department of Master of Computer Applications at G. Pulla Reddy Engineering College, Kurnool, Andhra Pradesh, India. She presented two research papers in national conferences. A paper was published in International Journal(IRECOS). Her research areas include Computer Networks and Routing Algorithms.

G. Raghuram graduated from Sri Krishna Devaraya University in the year 1994, MCA from Osmania University in the year. He is presently Associate Professor in the Department of Mastaer of Computer Applications at G. Pulla Reddy Engineering College, Kurnool, Andhra Pradesh, India. He presented two research papers in national conferences. His research areas include Artificial Intelligence and Computer Networks.

Dr. K.E. Sreenivasa Murthy obtained B.Tech and M.Tech degrees in Electronics and Communication Engineering from Sri Venkateswara University, Tirupati, India in 1989 and 1992 respectively and Ph.D degree from Sri Krishna Devaraya University, Anantapur, India, in 1997. He presented more than 10 research papers in various national and international conferences and journals. He is at present working as principal at G. Pullaiah College of Engineering and Technology, Kurnool, India. His research interests include FPGA and DSP applications.

B. Gurunath Reddy graduated B.Sc from Sri Venkateswara University, Tirupati, India in 2005 and MCA from Sri Krishna Devaraya University, Anantapur, India in 2008. He is at present working as software trainee in Keane India Limited., India. His research interests include Artificial intelligence and Computer networks.

www.icgst.com

MSP430 and nRF24L01 based Wireless Sensor Network Design with Adaptive power control

S. S. Sonavane[1], V. Kumar[1], B. P. Patil[2]
[1]Department of Electronics & Instrumentation Indian School of Mines University, Dhanbad-826004, India
sssonavane@gmail.com, vkumar52@hotmail.com
[2] Department of Electronics Engineering Maharashtra Academy of Engineering, Alandi (Pune)
bpatil@maepune.com

Abstract
In this paper, we have developed a Low cost and low power Wireless Sensor Networks (WSNs) Node using MSP430 and Nordic nRF24L01. The architectural and circuit details are presented. This architecture fulfils the requirements of low power, compact size and self-organization with a new feature of adaptive Power Control. For Low power consumption Adaptive Power control technique is used. In this technique we can vary the transmitted power according to the distance between the nodes, which is also the different feature of this WSN. Adaptive power algorithm that uses both RF output Power and Transmission rate to be adjusted according to the distance between the Nodes which will maximize the battery life time. All the Radio modules available in the market are utilizing constant power transmission during its operation. Hence significant reduction in energy consumption is possible based on the proposed approach which prolongs the battery lifetime.

Keywords: *Wireless Sensor Networks (WSNs), MSP430, Nordic nRF24l01, Adaptive power algorithm.*

1. Introduction

A wireless sensor network is a network made up of hundreds or thousands of Sensor nodes, which are densely deployed in an unattended environment. These nodes are capable of communicating by means of wireless communications, sensing and self-computation (software, hardware, algorithms) [1]. Hence the wireless sensor network is the result of the combination of sensor, embedded techniques, distributed information processing, and communication mechanisms. The sensor network is more application specific than traditional networks designed to accommodate various applications. The organization and architecture of a sensor network should be designed or adapted to suit a special task so as to optimize the system performance, maximize the operation lifetime and minimize the cost. Thus, in order to maximize the sensor network lifetime, the sensor network architecture will most likely tip toward a localized approach.

We describe the design and implementation of a sensor node that utilizes emerging hardware, low cost components and new techniques to achieve high data rate, extremely low power operation. Low power operation is achieved not only through selection of efficient hardware, but also through low duty cycling and by Adaptive power algorithm implementation. One cycle of sleep, wakeup, and run is typically the cost of acquiring a single set of sensor samples. For the majority of the time the node is sleeping. While asleep, the microcontroller must maintain its state, while consuming little power and shutting down or disconnecting all peripherals including the radio. [2]
For our WSN node design, we chose the Texas Instruments MSP430 microcontroller. The MSP430 consumes only 2 microwatts in sleep mode while maintaining RAM. The collection of various features has been integrated to create the highest data rate, lowest power mote to date. The remainder of the paper is organized as follows: Section (2) focuses on hardware details of WSN Node. Section (3) emphasizes on Adaptive power algorithm using variable power and transmission rate. Section (4) discusses lifetime improvement using adaptive power algorithm. Section (5) concludes the paper.

2. Hardware details of WSN Node

The hardware consists of MSP430 connected with nRF24L01 as shown in Figure1.

2.1 MSP430F1612

The Texas Instruments MSP430 family of ultra low power microcontroller consists of several devices featuring different sets of peripherals targeted for various applications. The architecture, combined with five low power modes is optimized to achieve extended battery life in portable measurement applications. The device features a powerful 16-bit RISC CPU, 16-bit registers, and constant generators that attribute to maximum code efficiency [3].

2.2 NORDIC nRF24L01:

The nRF24L01 is a single chip radio transceiver for the global, license-free 2.4 GHz ISM band. The low cost nRF24L01 is designed to merge very high speed communications (up to 2Mbit/s) with extremely low power (the RX current is just 12.5mA) [4]. The transceiver consists of a fully integrated frequency synthesizer, a power amplifier, crystal oscillator, demodulator, modulator and Enhanced ShockBurst protocol engine. In addition, the nRF24L01 also offers an innovative on-chip hardware solution called 'MultiCeiver' that can support up to six simultaneously communicating wireless devices. This makes it ideal for building wireless Personal Area Networks in a wide range of applications. The PCB of this WSN node is circular, having two inches diameter.

Output power, frequency channels, and protocol set-up are easily programmable through an SPI-bus. Current consumption is very low, only 8.5mA at an output power of -6dBm and 12.5mA in RX mode. Built-in modes such as Power Down (400nA current) and Standby (32µA at 130µs wakeup), makes significant power savings easily realizable. The data rate can be chosen between 1 and 2Mbit/s. This allows for short time-on-air and therefore low power consumption [5].

3. Adaptive Power Control Algorithm

Power efficiency is very important in wireless sensor networks because the sensors typically run on batteries and long lifetime is highly desirable. The algorithm used here is to set the transmission strength of the route update message. By setting the transmission strength, nodes can store the RSS (Received Signal Strength) and with the known transmission setting, distance to the transmitter can be estimated the node. Based on the estimated distance the node will then adjust its transmission power. Variable Transmission Control sets the transmission power based specifically on the RSS. It has multiple limits with each being associated with a different transmission setting.

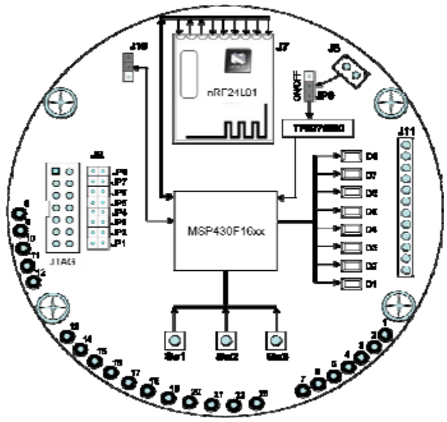

Figure 1 Front side of MSP430 and nRF24L01 based WSN node

The nRF24L01uses a Received Signal Strength which outputs an analogue signal which is proportional to the input signal level on the RFI/O. This output current is converted into a voltage by a 50 ohm resistor which is in turn is read by the Analog to Digital Converter (ADC) channel 0 of the microcontroller.

Whichever way these nodes are deployed or powered, the limited supply of power of each node will always be an issue. The radio transmitters and receivers of the nodes are usually the main consumers of the power supply. The two biggest current consumers are when the node transmits at maximum power (11.3 mA) or when the node

The Table-1 shows the settings used to obtain Adaptive power control in WSNs, which decides the power, according to the distance between the nodes. This research gives an Adaptive power based WSN node rather than current techniques, which supports constant output power.

The PA control is used to set the output power from the nRF24L01 power amplifier (PA). In TX mode PA control has four programmable steps, see Table1. The PA control is set by the RF_PWR bits in the RF_SETUP register.

This algorithm is developed to minimize the power consumption of the WSN node by the use of variable power and variable transmission Rate. Previous method uses either Transmission Power or rate is to be varied according to the distance [6]. But we had developed a new algorithm with variable power and transmission rate both. The Received signal strength is considered to calculate the distance between the Nodes. The received signal strength is taken from RFI/O terminal of nRF24L01. This voltage is then fed to the internal ADC of MSP430.

SPI RF-SETUP (RF_PWR)	RF Output Power	DC Current Consumption
11	0 dBm	11.3 mA
10	-6 dBm	9.0 mA
01	-12 dBm	7.5 mA
00	-18 dBm	7.0 mA

Table-1 Output power control Settings at Nordic

The air data rate is the modulated signaling rate the nRF24L01 uses when transmitting and receiving data. The air data rate can be 1Mbps or 2Mbps. The 1Mbps data rate gives 3dB better receiver sensitivity compared to 2Mbps. High air data rate means lower average current consumption and reduced probability of on-air collisions. The air data rate is set by the RF_DR bit in the RF_SETUP register.

The Data Rate is kept at 2Mbps when the node is transmitting with higher power levels (i.e. at 0dBm and -6dBm) and 1 Mbps when transmitting at low power levels (i.e. at -12dBm and -18dBm) as given in Table-2.

Distance Between Nodes (Meter)	Output Power Setting at Transmitter	Transmission Rate (Mbps)
8 to 10	0 dBm	2
6 to 8	-6 dBm	2
3 to 6	-12 dBm	1
0 to 3	-18 dBm	1

Table-2 output Transmission Rate Settings

We had tested the nodes for a distance of 10 meters. The Power consumption will change as per the distance as shown in Table-3. The flow chart for adaptive algorithm is shown Figure2. The algorithm is developed such that if the nodes are close to each other the transmission power and data rate is adjusted at lower level. As the distance increases, the transmission power and data rate goes on increasing.

4. Experimental results:-

The WSN nodes are arranged in the network as shown in Figure 3 where N1, N2, N3, N4, N5, N6 are the different Nodes. 'S' indicates the destination node which is connected to computer. The data from the network is processed and displayed by this computer. The Node lifetime depends upon the power consumption of MSP430, transmit and receive mode consumption of nRF24L01. Also it depends upon the sleep and active times i.e. duty cycle [7,8]. The node is programmed with <1% of duty cycle [9]. The Average current consumption and the node lifetime are calculated by taking the different parameters as shown in Table 4.

Distance Between Nodes (Meter)	RF Output Power	DC Current Consumption	RF Output mV	Settings At PWR_UP Register	Output Rate Adjustment (Mbps)
8 to 10	0 dBm	11.3 mA	56.5	11	2
6 to 8	-6 dBm	9.0 mA	45	10	2
3 to 6	-12 dBm	7.5 mA	37.5	01	1
0 to 3	-18 dBm	7.0 mA	35	00	1

Table-3 Output power and rate settings for Adaptive power control Algorithm

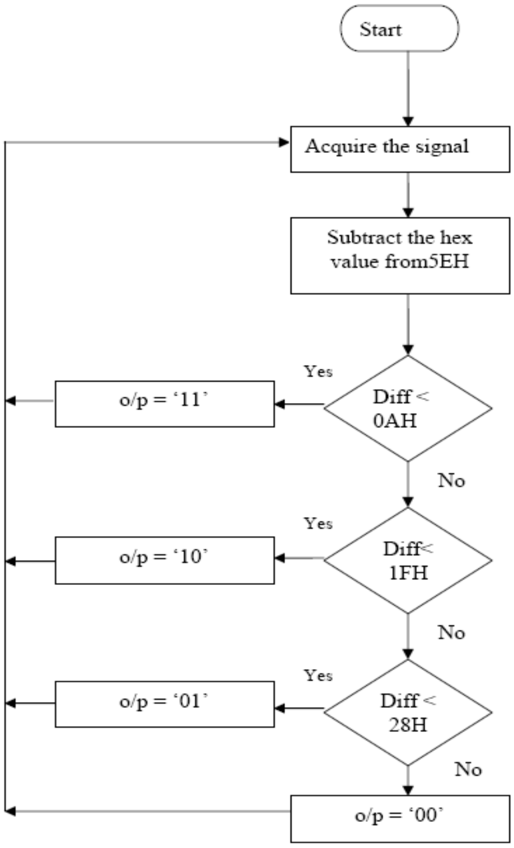

Fig. 2 Adaptive power control Algorithm Flow chart

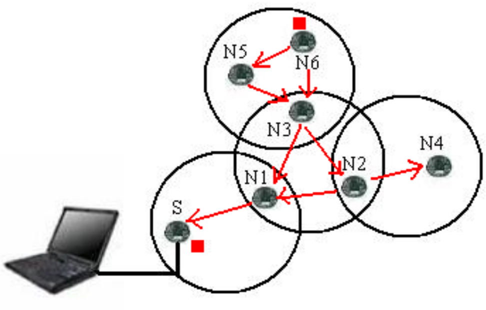

Figure 3 Intercommunication between the nodes

As Node 1 is very close to destination 'S', instead of constant power of 0 dbm the node can be set to lowest power at -18 dbm. This will increase battery lifetime from 4.2 years to 7 Years as shown in Figure 4.

This will apply to all other nodes. Each will set its power to different levels depending upon the distance between Node and destination Node 'S'.

Parameter	Settings	Unit
Overhead	65	bits
Payload length	8	bits
Packet length	73	bits
Bit rate	2000000	bits/sec
Time on air	0.3	sec
Time in RX	0.00003252	sec
MCU+TX Current	11.6	mA
MCU+RX current	12.9	mA
PLL- Lock time	0.00013	sec
PLL- Lock TX current	8	mA
PLL- Lock RX current	8.4	mA
Power_Dn current	0.0009	mA
Duty-cycle period	55.7	sec
Power_up current	0.285	mA
Power_up time	0.0015	sec
I (avg)	0.06342619	mA
Battery Rating	2450	mAh
Lifetime	38627.5768	hours
	1609.48237	days
	4.42165486	**years**

Table- 4 Node lifetime calculations for 0dBm power

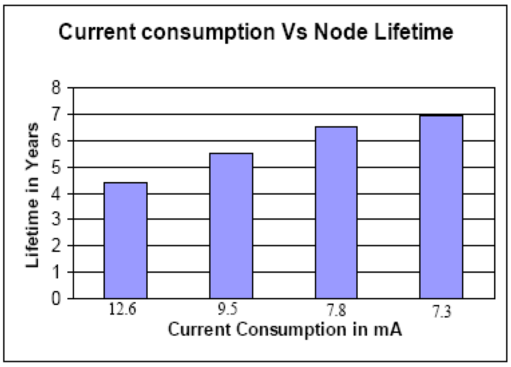

Figure 4 Node lifetime for various transmitter levels of nRF24L01

The WSN Node designed Cost Details for one Node is given in Table5:

Sr. No	Name of the Component	Cost
1.	MSP430F2013	0.5 $
2.	nRf24L01	1.2 $
3.	Crystal	0.1 $
4.	Antenna & other Components	0.3 $
5.	PCB Development	0.1 $
6.	Battery	0.3 $
	Total Cost	**2.4 $**

Table-5 Cost Analysis of the WSN node

If we manufacture the nodes in bulk, then the cost will be reduced around 1$. Thus we get a cost effective solution through this design for various applications in automobiles also [10].

5. Conclusion

We have designed a very compact node with maximum possible power efficiency. The node has many ports available for future expansion of the nodes. The Power management occurs as Low duty cycle is considered. We have developed our own protocol stack efficient than Zigbee with simple programming compilers like cross studio.

The adaptive algorithm saves the power to a great extent as at high power level transmission due to high data rate the transmission will be faster causing the node ON time to be reduced. The transmission time is almost half than required by 1 Mbps rate. Hence the battery life can be further increased.

6. References

[1] Akyildiz, W. Su, Y. Sankarasubramaniam and E. Cayirci. A survey on Sensor Networks. IEEE Communications Magazine, vol. 40, Issue: 8, pp. 102-114, August 2002.

[2] S. S. Sonavane, V. Kumar and B. P. Patil. Design Factors, Requirements and Research issues of wireless Sensor Networks. International journal of Engineering Research & Industrial Applications (IJERIA), Vol.1, No. III, pp 79-93, 2008.

[3] http://www.ti.com/msp430/datasheet accessed on 20th August 2008.

[4] http://www.nordicsemiconductor.com/nrf24L01 accessed on 10th sept. 2008.

[5] S. S. Sonavane, V. Kumar and B. P. Patil. Component Choice for Low Power Wireless Sensor Networks Node. International Journal of Computer, Information Technology and Engineering (IJCITAE), Vol.2, No.1, July-2008, Series Publications.

[6] D. Estrin, D. Culler, K. Pister and G. Sukhatme. Connecting the physical world with pervasive networks. IEEE Pervasive Computing, 1(1), Jan- 2002.

[7] S. S. Sonavane, B. P. Patil and V. Kumar. Designing Wireless Sensor Network with Low cost and Low Power. International Conferences On Networking (ICON) to be held at IIT Roorki on 12th Dec-08.

[8] S. S. Sonavane, B. P. Patil, V. Kumar. Energy Efficient protocols Wireless Sensor Network: an Overview. In proceedings of International Conference on Computer and Communication Engineering (ICCCE08) at Department of Electrical and Computer Engineering, International Islamic University, Malaysia (IIUM), May-2008.

[9] Joseph Polastre, Robert Szewczyk, and David Culler. Telos: Enabling ultra-low power wireless research. The Fourth International Conference on Information Processing in Sensor Networks: Special track on Platform Tools and Design Methods for Network Embedded Sensors (IPSN/SPOTS), Los Angeles, California, April 2005.

[10] Fred Yu, Bozena Kaminska and Pawel Gburzynski. A Wireless Sensor-Based Driving Assistant for Automobiles. ICGST-ACSE Journal, ISSN 1687-4811, Volume 8, Issue II, December 2008.

7. Biographies

Sonavane S. S. is currently pursuing his Ph.D. from Indian School of mines, Dhanbad. He is currently working as Assistant Professor in Dept. of Electronics and Telecommunication Engg. in Rajarshi Shahu College of Engineering, Pune. He has published around 18 papers in International and National journals and Conferences.

Dr. V. Kumar received Ph.D in 1980 and worked as Scientist at CMRI, Dhanbad and Visiting Scientist at CNR, Frascati, Roma, Italy during 1983-84. He has attended number of International Conferences including conferences held in USA, Portugal, Singapore and Italy. At present, he is Associate Professor and Head of Electronic & Instrumentation Department, ISM University, Dhanbad. He has published over 100-research papers; guided 2-Ph.D. and number of M. Tech students in the areas of Opto-electronic materials and Optical Fiber Sensors.

Dr. B. P. Patil received his Ph.D. in 2000 and has more than 18 years of teaching and industrial experience. He has published 41 papers in International journals and Conferences also guiding two Ph.D. students. He is currently working as Professor and Head of Electronics Engineering Dept. at Maharashtra Academy of Engineering (MAE), Alandi, and Pune. His area of research includes sensor network, wireless communication.

www.icgst.com

Design and Reliability Analysis of a new Fault-tolerant Multistage Interconnection Network

Rinkle Aggarwal, Lakhwinder Kaur, Himanshu Aggarwal

Department of Computer Science & Engineering, Thapar University, Patiala –147004 (India)
raggarwal@thapar.edu
Department of Computer Engineering, Punjabi University, Patiala–147002 (India)
mahal2k8@yahoo.com, himagrawal@live.com

Abstract

The design of a suitable interconnection network for inter-processor communication is one of the key issues of the system performance. The reliability of these networks and their ability to continue operating despite failures are major concerns in determining the overall system performance. In this paper a new irregular network IASEN (Irregular Augmented Shuffle Exchange Network) has been proposed, which is derived from the regular ASEN-2 (Augmented Shuffle Exchange Network). ASEN-2 is a multipath network with limited fault tolerance. The reliabilities of the IASEN and ASEN-2 multi-stage interconnection networks have been calculated and compared in terms of the Upper and Lower bounds of Mean Time To Failure (MTTF). It has been observed that the proposed IASEN multistage interconnection network provides much better fault-tolerance by providing more paths between any pair of source-destination and better reliability at the expanse of little more cost than ASEN-2.

Keywords: *Multistage Interconnection Networks, Reliability, Augmented Shuffle Exchange Network, Irregular Augmented Shuffle Exchange Network, Fault-tolerance.*

1. Introduction

Advances in LSI and VLSI technology are encouraging greater use of multiple-processor systems with processing elements to provide computational parallelism and memory modules to store the data required by the processing elements. Interconnection Networks (INs) play a major role in the performance of modem parallel computers. Many aspects of INs, such as implementation complexity, routing algorithms, performance evaluation, fault-tolerance, and reliability have been the subjects of research over the years. There are many factors that may affect the choice of appropriate interconnection network for the underlying parallel computing environment [8,9]. Though crossbar is the ideal IN for shared memory multiprocessor, where N inputs can simultaneously get connected to N outputs, but the hardware cost grows astronomically. Multistage Interconnection Networks (MINs) are recognized as cost-effective means to provide programmable data paths between functional modules in multiprocessor systems. These networks are usually implemented with simple modular switches, employing two-input two-output switching elements. Most of the MINs proposed in the literature have been constructed with 2x2 crossbar switches as basic elements, and have $n = \log_2 N$ switching stages with each stage consisting of N/2 elements, which makes the cost of this network as O(N logN), as compared to $O(N^2)$ for a crossbar [6]. The pattern of interconnection may be uniform or non-uniform, which classifies the MINs to be regular or irregular respectively[10]. In the case of irregular networks, the path length varies from any input to any output, in contrast with regular networks, where it is the same [13]. Fault-tolerance in an interconnection network is very important for its continuous operation over a relatively long period of time [11]. Many networks have been designed and proposed to increase the fault-tolerance in the literature [2,3,9,13,15,16,17]. Permutation capability and other issues related to routing have also been extensively researched [2,5,15,16,17]. Bandwidth and other performance parameters have also been studied in-depth [12,14]. However, little attention has been paid to the computation of reliability of these networks. Reliability is measured in terms of Mean Time to Failure (MTTF), which is evaluated using simple series-parallel probabilistic combinations. This analysis is based upon the lower and upper bounds of the network reliability. This paper has been organized into five sections whose details are as follows.

Section 1 introduces the subject under study. Section 2 describes the structure and design of networks. Section 3 focuses on the reliability analysis. Section 4 concentrates on the reliability analysis of IASEN. Section 5 describes the cost effectiveness. Finally, the conclusion has been presented.

2. Structure and design of networks

2.1 ASEN-2 Network

Augmented Shuffle Exchange Network (ASEN-2) is a regular network, having equal number of switches in each of the stage. ASEN-2 network is constructed from Shuffle Exchange Network by adding a stage of 2×1 multiplexers at the initial stage and 1×2 demultiplexers at last stage. It provides multiple paths between a source and a destination. ASEN-2 of size N × N with N number of sources and N number of destination consists of $\log_2 N$ -1 stages where the initial stage consists of N/2 switches of size 3×3 and the last stage consist of N/2 switches of size 2×2. ASEN-2 provides fault tolerance using links between the conjugate pairs of switches [13]. A 16×16 ASEN-2 network is shown in Figure 1.

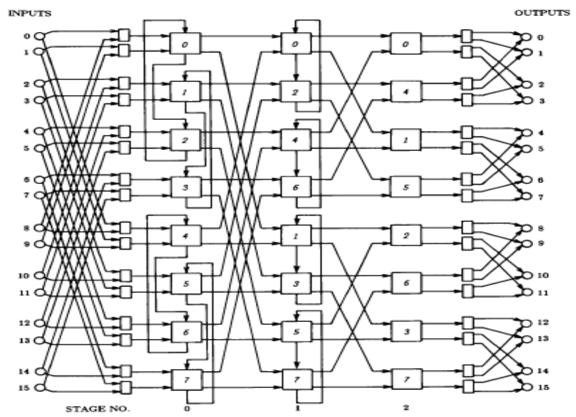

Figure 1. Augmented Shuffle Exchange network (ASEN-2)

2.2 Irregular ASEN

Irregular Augmented Shuffle Exchange Network (IASEN) shown in Figure 2 is derived from ASEN-2 multistage interconnection network. The switches in the first stage form a loop to provide multiple paths if a fault occur in the next stage. Each source is connected to two different switches in each group with the help of multiplexer and each destination is connected with demultiplexer. Following structural changes have been made in IASEN in comparison to ASEN-2.

1) Four switches removed from the stage 1 (Intermediate Stage)
2) Use of 1 x 4 DEMUX in place of 1 x 2 DEMUX
3) Loops and connections changed

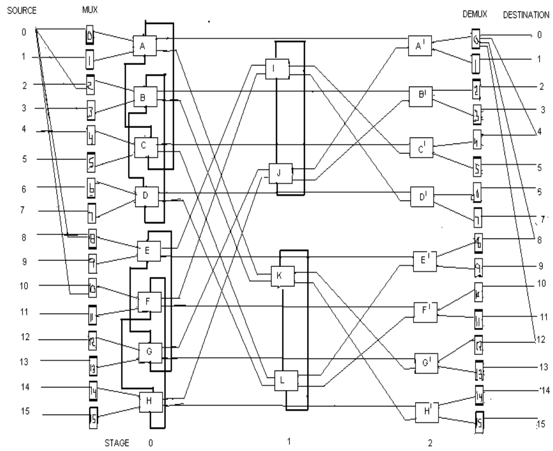

Figure 2. Irregular Augmented Shuffle Exchange Network

3. Reliability Analysis

The reliability analysis of hardware [13] and software[1] are two very important considerations. The reliability analysis here focuses on the hardware reliability of the networks. Reliability of ASEN-2, IASEN networks are analyzed in terms of Mean time to Failure (MTTF). MTTF of a MIN is evaluated using simple series-parallel probabilistic combinations.

3.1 Series configuration model

A series configuration model is constructed by connecting all the components in a series system. This type of configuration is very sensitive because the failure of a single component make whole of the system fail. The reliability of a series configuration model is always worse than the poorest component in it. Series configuration model is shown in Figure 3.

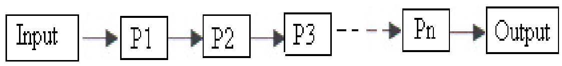

Figure 3. Series configuration model

3.2 Parallel configuration model

In this all components are connected parallel to each other. In this if one component fails than the data can follow another path and system will be active. This type of model is shown in Figure 4.

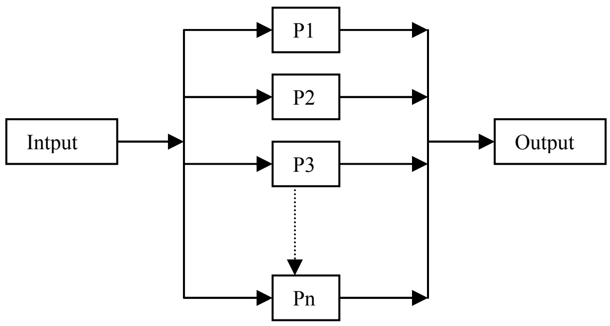

Figure 4. Parallel configuration model

3.3 Series-Parallel Configuration model

The two loops in a conjugate pair are in parallel and all the conjugate pairs of loops are in series as shown in Figure 5.

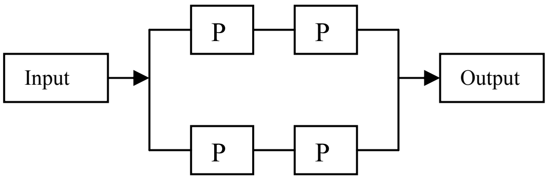

Figure 5. Series-Parallel configuration model

To make the Reliability analysis traceable, we need to have some assumptions. We use the assumptions similar to the ones that have been made previously in the other studies of fault tolerant networks [3,4,9,10]. The assumptions used in the analysis on the failure rates of the components are given below:

1. Switch failure occur independently in a network with a failure rate of λ for 2×2 crossbar switches (a reasonable estimate for λ is about 10^{-6}per hour).
2. Failure of the multiplexers and demultiplexers also occur independently with failure rates of λ_m and λ_d respectively.
3. Assuming that the hardware complexity of a component is directly proportional to the gate counts of it, one can derive a failure rate of the component From the basic logic design of MUX and DEMUX, we can say that number of gates in a 2m × l MUX or a 1 × 2m DEMUX is roughly double of that in a m × l MUX or a 1 × m DEMUX. Based on the gate counts of crossbar switches, the number of gates in a 2 × 2 crossbar switch is approximately equal to that in a 2 × 1 MUX or a 1 × 2 DEMUX. Thus to simplify the analysis we can assume that $\lambda_m = m\lambda/2$ for a m × 1 MUX, where λ_m failure rate of MUX or λ_d ($=\lambda_m$) for 1 × m DEMUX, where λ_d failure rate of DEMUX.

   The adaptive routing scheme considers a 2 × 2 switch in the last stage and its associated DEMUX as a series system, so we consider these three elements as single component (SE2d), and based on a gate count, a failure rate of $\lambda_{2d} = 2\lambda$ can be assigned to this group of elements. Also let λ_2 and λ_3 be the failure rate for the 2 × 2 (SE2) and the 3 × 3 switch (SE3), then based on gate count, $\lambda_2 = \lambda$ and $\lambda_3 = 2.25\lambda$ and $\lambda_{3m} = 4.25\lambda$.
4. Irregular MINs are inherently multi-path and the MTTF needs to be calculated at all existing path-lengths separately based upon the series and parallel models of reliability.

ASEN-2 Reliability

For the reliability both the optimistic and pessimistic analysis of the networks has been done. These were extended to incorporate the added complexity of the switches used in the ASEN-2, the multiplexers and demultiplexers used at the input and output interfaces of the network.

3.4.1 ASEN-2 Optimistic (Upper bound) Analysis

To obtain an upper bound for the ASEN-2 and observed that each source is connected to two multiplexers and each switch is connected is a conjugate pair. So if we assume that the ASEN-2 is operational as long as one of the two multiplexers attached to a source is operational and both components in a conjugate pair are not faulty [10]. Block diagram of upper bound is shown in Figure 6.

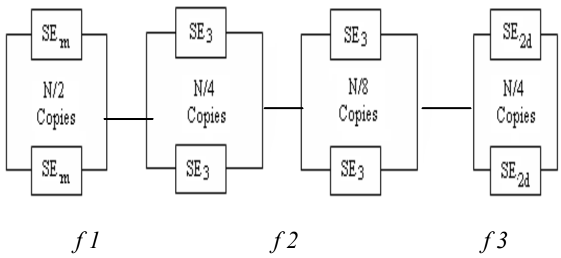

Figure 6. Upper Bound of ASEN-2

Reliability Equations are:

$$f1 = \left[1 - \left(1 - e^{-\lambda_m t}\right)^2\right]^{\left(N/2\right)}$$

$$f2 = \left[1 - \left(1 - e^{-\lambda_3 t}\right)^2\right]^{\left(N/4 + N/4 (n-3)\right)}$$

$$f3 = \left[1 - \left(1 - e^{-\lambda_{2d} t}\right)^2\right]^{\left(N/4\right)}$$

$$R_{Optimistic} = f1 * f2 * f3$$

$$MTTF = \int_0^{\infty} R_{Optimistic}(t)\, dt$$

3.4.2 ASEN-2 Pessimistic (Lower Bound) Analysis

At the input side of the ASEN-2, the routing algorithm does not consider the multiplexers to be an integral part of a given 3 × 3 SE. For example, as long as one of the two multiplexers attached to switch 0 is operational, switch 0 can still be used for routing. Hence, if we group two multiplexers with each switch on the input side and consider them as a series system (SE_{3m}), then we will have a conservative estimate of the reliability of these three components. Their aggregated failure rate will be λ_{3m}= 4.25λ. Finally, these aggregated components and the switches in the intermediate stages can be arranged in pairs of conjugate loops. To obtain lower bound of the ASEN-2, assume the network is failed whenever more than one loop has a faulty element or more than one switch in the last stage fails [10]. The block diagram is shown in Figure 7.

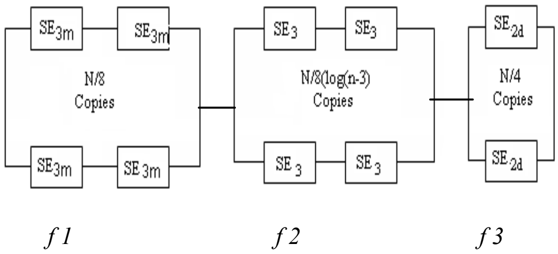

Figure 7. Lower Bound of ASEN-2

Reliability Equations are:

$$f1 = \left[1 - \left(1 - e^{-\lambda_{3m} t}\right)^2\right]^{\left(N/4\right)}$$

$$f2 = \left[1 - \left(1 - e^{-\lambda_3 t}\right)^2\right]^{\left(N/4\,(n-3)\right)}$$

$$f3 = \left[1 - \left(1 - e^{-\lambda_{2d} t}\right)^2\right]^{\left(N/4\right)}$$

$$R_{Pessimistic} = f1 * f2 * f3$$

$$MTTF = \int_0^{\infty} R_{Pessimistic}(t)\, dt$$

4. Reliability Analysis of IASEN

4.1 IASEN Optimistic (Upper bound) Analysis

Upper bound formula of IASEN and MASEN is same. Only difference is the value of λ_{2d}, for MASEN its value is 2λ and for IASEN its 3λ. Difference in this value is due to the same reason that for MASEN λ_{2d} means one 2 × 2 switch and two 1 × 2 demultiplexers (λ+λ=2λ), whereas for IASEN λ_{2d} means one 2 × 2 switch and two 1 × 4 demultiplexers (λ+2λ=3λ). The block diagram is shown in Figure 8.

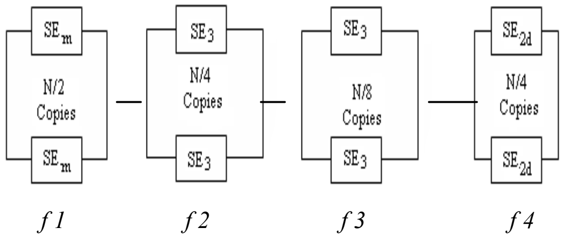

Figure 8. Block diagram of Upper Bound IASEN

Reliability Equations are:

$$f1 = \left[1 - \left(1 - e^{-\lambda_m t}\right)^2\right]^{\left(N/2\right)}$$

$$f2 = \left[1 - \left(1 - e^{-\lambda_3 t}\right)^2\right]^{\left(N/4\right)}$$

$$f3 = \left[1 - \left(1 - e^{-\lambda_3 t}\right)^2\right]^{\left(N/8\right)}$$

$$f4 = \left[1 - \left(1 - e^{-\lambda_{2d} t}\right)^2\right]^{\left(N/4\right)}$$

$$R_{Optimistic} = f1 * f2 * f3 * f4$$

$$MTTF = \int_0^{\infty} R_{Optimistic}(t)\, dt$$

4.2 IASEN Pessimistic (Lower bound) Analysis

For the lower bound, each group is considered independently and is assumed to be faulty if there is any single fault in it. Lower bound equation of IASEN and MASEN is also same. Only difference is again in the value of λ_{2d}, for MASEN its value is 2λ and for IASEN its 3λ. Difference in this value is due to the reason that for MASEN λ_{2d} means one 2 × 2 switch and two 1 × 2 demultiplexers (λ+λ=2λ), whereas for IASEN λ_{2d} means one 2 × 2 switch and two 1 × 4 demultiplexers (λ+2λ=3λ). The block diagram is shown in Figure 9.

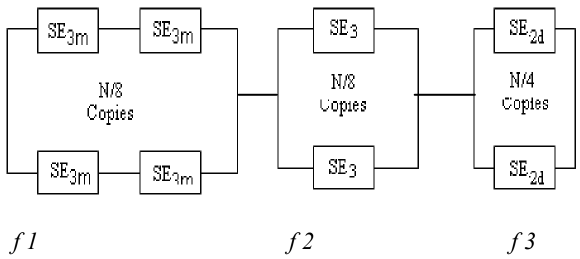

Figure 9. Block diagram of Lower Bound IASEN

Reliability Equations are:

$$f1 = \left[1 - \left(1 - e^{-\lambda_{3m} t}\right)^2\right]^{\left(N/8\right)}$$

$$f2 = \left[1 - \left(1 - e^{-\lambda_{3} t}\right)^2\right]^{\left(N/8\right)}$$

$$f3 = \left[1 - \left(1 - e^{-\lambda_{2d} t}\right)^2\right]^{\left(N/4\right)}$$

$$R_{Pessimistic} = f1 * f2 * f3$$

$$MTTF = \int_0^{\infty} R_{Pessimistic}(t)\, dt$$

The Reliability equations are solved using Trapezoidal rule of integration and the results are shown in Table 1. Based on Table 1 a comparison of Reliability values of ASEN-2 and IASEN in terms of Lower and Upper-bounds of MTTF for various network sizes has been shown in Figures 10 and 11 respectively.

Network Size (LogN)	ASEN-2		IASEN	
	(LB)	(UB)	(LB)	(UB)
4	5.07329	5.13012	5.92568	5.82242
5	4.84479	4.89034	5.7212	5.6301
6	4.62714	4.67522	5.52521	5.44324
7	4.44248	4.47502	5.33551	5.2603
8	4.25612	4.28454	5.15041	5.08016
9	4.07555	4.10075	4.96866	4.90202
10	3.89916	3.92184	4.78934	4.72529

Table 1: MTTF values for different network size

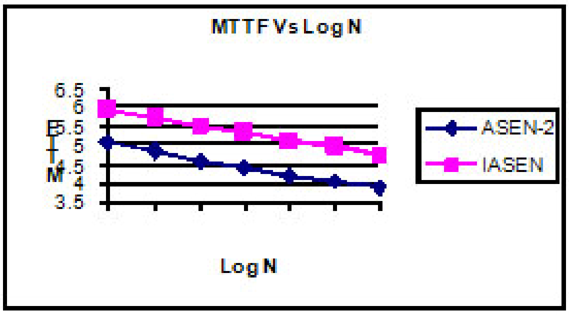

Figure 10. Lower Bound MTTF Comparison

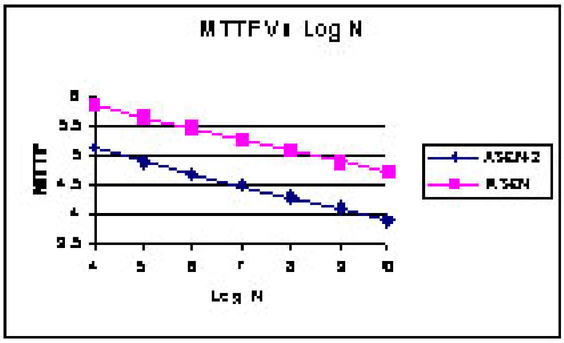

Figure 11. Upper Bound MTTF Comparison

5. Cost Effectiveness

To estimate the cost of a network the assumption is made that the cost of a switch is proportional to the number of cross-points within a switch. For example a 4x4 switch has 16 units of hardware cost whereas a 2x2 switch has 4 units [1,8,11,14,15,16]. The cost functions for ASEN-2 and IASEN are given in the table 2. Table 3 shows the data values of cost functions. The graph for these data values is shown in Figure 12.

Network	Cost
ASEN-2	$3N(1.5 \log_2 N - 1)$
IASEN	$3N(1.5 \log_2 N) - 20$

Table 2. Cost Functions

Network Size (Log N)	Cost	
	ASEN-2	IASEN
4	7.90	8.06
5	9.17	9.35
6	10.58	10.74
7	11.75	11.89
8	13.04	13.16
9	14.17	14.28
10	15.39	15.49

Table 3. Cost values for different network size

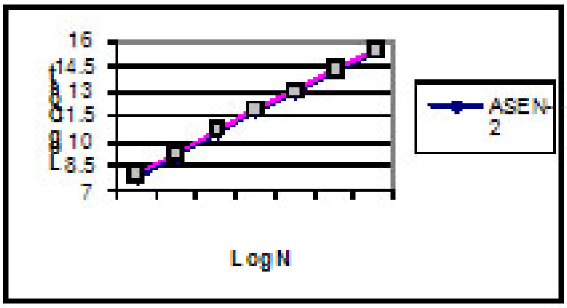

Figure 12. Cost comparison of ASEN-2 and IASEN

6. Conclusion

An Irregular (IASEN) is designed using existing regular Augmented Baseline Network (ASEN-2) It has comparatively one more stage. IASEN is a dynamically re-routable and provides multiple paths of varying lengths between a source-destination pair. It has been found that in an IASEN, there are eight possible paths between any source-destination pair, whereas ASEN-2 has only six such paths. The reliability analysis shows that IASEN has better performance than ASEN-2. Thus the new network IASEN provides better fault-tolerance and reliability than the existing ASEN-2 with little more cost.

7. Glossary

Reliability: Reliability of a system is the probability that it will perform its intended function satisfactorily for a given time under stated operating conditions.

MTTF: The MTTF of a MIN is defined as the expected time elapsed before some source is disconnected from some destination.

Fault-tolerance: The ability of the network to operate even in the presence of faults, although at a degraded performance.

Bandwidth: It is defined as mean number of active memory modules in a transfer cycle of the MIN. It also takes into account the memory access conflicts caused by random nature of processor requests.

Permutation: Permutation passibility of a network shows that at a particular moment of time, if a number of requests simultaneously occur at source, how many of them will successfully mature i.e. reach the destination.

8. References

[1] A. Sheta, Parameter Estimation of Software Reliability Growth Models by Particle Swarm Optimization, ICGST-AIML Journal, Vol. 7, No.7, June,2007, pp.55-61.

[2] H. Aggarwal, and P.K. Bansal, Routing and Path Length Algorithm for cost effective Modified Four Tree Network, IEEE TENCON, 2002, pp. 293-294.

[3] Harsh Sadawarti and P.K Bansal., "Fault tolerant Irregular Augmented Shuffle Network" WSEAS International Conference on Computer Engineering and Applications, Gold Coast, Australia, Jan, 2007, pp. 17-19.

[4] J. Sengupta and P.K Bansal, "Reliabilty and performance measures of regular and irregular Multistage Interconnection Networks", Interational Conference IEEE TENCON 2000, pp. 531-536.

[5] J. Sengupta and P.K. Bansal, "Performance Analysis of Regular and Irregular Dynamic MINs", International Conference IEEE TENCON 99, Sept. 1999, Cheju Island, Korea, pp. 427-430.

[6] Jose Duato, Sudhakar Yalamanchili and Ni Lionel, "Interconnection Networks: An Engineering Approach", IEEE Computer Society, 1997.

[7] M. Lubazewski and B Coutois, "A Reliable Fail-safe System", Parallel and Distributed Systems, IEEE Computer Society, Vol. 47, No. 2, Feb. 1998, pp. 236-241.

[8] N Laxmi Bhuyan, Yang Qing and Dharma P Agrawal., "Performance of Multiprocessor Interconnection Networks", IEEE Computer, Vol. 22, Feb. 1993, pp. 25-37.

[9] Nitin, "On analytic Bounds of Regular and Irregular Fault-tolerant Multistage Interconnection Networks", International Conference on PDPTA, June 2006, pp. 26-29.

[10] P. K. Bansal, R.C. Joshi and K. Singh, "On a Fault-tolerant Multi-stage Interconnection Network", International journal of Electronics and Electrical Engineering, Vol. 20, No.4, 1994, pp. 335-345.

[11] P. K. Bansal, R.C. Joshi, K. Singh K. and G. P. Siroha, "Fault-tolerant Augmented Baseline Multistage Interconnection Network", Proc. International Conference IEEE TENCON 91, India, Aug. 1991, pp. 200-204.

[12] R. Aggarwal, H. Aggarwal and L. Kaur, "On Bandwidth analysis of Irregular Fault-tolerant Multistage Interconnection networks", International Review on Computers and Software, Vol. 3, No. 2, March 2008, pp. 199-202.

[13] R. Aggarwal and L. Kaur, **"**On Reliability Analysis of Fault-tolerant Multistage Interconnection Networks**"**, International Journal of Computer Science and Security**,** Vol. 2, No. 4, August 2008, pp. 1-8.

[14] R. Aggarwal, H. Aggarwal and L. Kaur, **"**Performance Analysis of of Regular and Irregular Multistage Interconnection Networks**"**, International Journal of Information and Computing Science**,** Vol. 11, No. 2, December 2008.

[15] Sandeep Sharma, K.S. Kalhon and P.K. Bansal, "On a class of Multistage Interconnection Networks in parallel processing" IJCSNS International Journal of Computer Science and Network Security, Vol.8, No.5, May 2008, pp. 287-291.
[16] Sandeep Sharma, K.S. Kalhon, P.K. Bansal and Kawaljeet Singh, "Improved Irregular Augmented Shuffle Exchange Multistage Interconnection Network", International Journal of Computer Science and Security, Vol. 2, No. 3, 2008, pp. 28-33.
[17] Sandeep Sharma, K.S. Kalhon, P.K. Bansal and Kawaljeet Singh, "Irregular class of Multistage Interconnection Networks in Parallel Processing", Journal of Computer Science, Vol. 4, No. 3, 2008, pp. 220-224.

Biographies

Rinkle Aggarwal, B.Tech (Computer Science & Engg.), M.S. (Software Systems), is Senior Lecturer in Computer Science & Engineering Department at Thapar University, Patiala. She has more than 11 years of teaching experience and served academic institutions such as Guru Nanak Dev Engineering College, Ludhiana and S.S.I.E.T 'Derabassi. She has supervised many M.Tech. Dissertations and contributed 15 articles in Conferences and 6 papers in research Journals. Her areas of interest are Parallel Computing and Algorithms.

Lakhwinder Kaur Ph.D. is Reader in Computer Engineering at University College of Engineering Punjabi University, Patiala. She has 16 years of teaching experience. She has published 8 research papers in International Journals. Her areas of interest are Image processing, Parallel Computing and Computer Graphics.

Himanshu Aggarwal, Ph.D., is Reader in Computer Engineering at University College of Engineering, Punjabi University, Patiala. He has more than 15 years of teaching experience and served academic institutions such as Thapar Institute of Engineering & Technology, Patiala, Guru Nanak Dev Engineering College, Ludhiana and Technical Teacher's Training Institute, Chandigarh. He is an active researcher who has supervised many M.Tech. Dissertations and contributed 30 articles in Conferences and 8 papers in research Journals. His areas of interest are Information Systems, ERP and Parallel Computing.

www.icgst.com

The Impact of Packet Size and Packet Dropping Probability on Bit Loss of VoIP Networks

HESHAM N. ELMAHDY and MOHAMED H. N. TAHA
Faculty of Computers and Information, Cairo University, Giza, Egypt
ehesham@mailer.eun.eg, mnasrtaha@gmail.com

Abstract
The demand for voice over IP (VoIP) applications has increased tremendously through the last two decades. This great demand leads to a great increase in the Quality of Service (QoS) researches and other related fields. One of these fields is the Differentiated Services (DiffServ). In this paper, we studied the effect of the packet size and the effect of random early detection (RED) parameters on the Two Rate Three Color Marker (trTCM) and Single Rate Three Color Marker (srTCM). This is done via a computer simulation using a network simulator (NS-2). Through this paper, we introduce a new simulation model. We will try through this model to find the most suitable parameters such as dropping probability and packet size, in order to achieve better fairness and better goodput. Beside that, we will introduce the standard deviation (SD) as another fairness measuring technique.

Keywords:
Quality of service (QoS); Differentiated Service; Traffic Marker; Random Early Detection (RED); Fairness index; Two Rate Three Color Marker (trTCM); Single Rate Three Color Marker (srTCM).

1. Introduction

Voice over IP (VoIP) uses the Internet Protocol (IP) to transmit voice as packets over an IP network. It is a set of technologies that enable voice calls to be carried over the Internet (or other networks designed for data), rather than the traditional telephone landline system—the Public Switched Telephone Network, or PSTN. In other words, it is a way to have telephone conversations with others using the Internet rather than a traditional telephone line. So VoIP can be achieved on any data network that uses IP, like Internet, Intranets and Local Area Networks (LAN). Here the voice signal is digitized, compressed and converted to IP packets and then transmitted over the IP network.[1] The potential for very low-cost or free voice calls is driving the use of the technology. But in the long-term, VoIP is more significant than just free phone calls; it represents a major change in telecommunications.

The increase in both popularity and capacity of the Internet has led to the increasing need to provide real-time voice and video services to the network. For most VoIP users the packet loss, delay and the fairness of service provided are common problems. However, the internet does not provide any QoS guarantees to voice applications. Many protocols have been developed for achieving the QoS over the internet. The Diffserv is one of these protocols. It was developed by IEFT to provide QoS to voice applications over the internet [2].

QoS refers to the general concept of prioritization network traffic. It is the ability of a network element to have some level of assurance that its traffic and service requirements can be satisfied. By default, each packet is treated equally and in a first-come, first-served basis. Certain traffic patterns can be given higher priority or can be guaranteed specific network resources by utilizing QoS. The main QoS problem in the Internet is how to provide and guarantee the bandwidth, delay and packet loss bounds to the real-time network traffic. Other QoS problems include end-to-end QoS deployment, network security, and network reliability [3].

VoIP can guarantee high-quality voice transmission only if the voice packets, for both the signaling and audio channel, are given priority over other kinds of network traffic. For VoIP to be deployed so that users receive an acceptable level of voice quality, VoIP traffic must be guaranteed certain compensating bandwidth, latency, and jitter requirements. QoS ensures that VoIP voice packets receive the preferential treatment they require.

The Diffserv provides an end to end QoS guarantee by enforcing each routing and switching node to perform different types of functions related to the QoS metrics. These metrics could be bandwidth, delay or packet loss. A DiffServ network achieves its goals by separating between the edge and core network. The edge network performs complex tasks such as traffic classification, traffic monitoring and traffic marking. The core network examines packets' code and forwards them accordingly. As a result, the core network will need to implement active queue management schemes in order to provide service

differentiation to the traffic [4]. This service differentiation will be according to service classes and dropping precedence. The major advantages of the Diffserv approach are that it is a good match to the Internet architecture and that it can be initially deployed with a minimalist approach, adding complexity as needed.
A lot of researches were done in order to measure and achieve better fairness in the provided services. Fairness index is used for this purpose. The Stander deviation is another measure which may be used for this purpose as well.
In this paper, we will focus on the relation between the SD of the lost bits and the change in one of the RED parameters. Beside that we will monitor the effect of the packet size on the stander deviation. The trTCM and srTCM will be the traffic policing schemas used in our simulations. We will introduce the SD as a new efficient measure for the network fairness. In addition, we will get a relation between the RED parameters and the SD. The rest of the paper is organized as follows. Section 2 gives a brief description of the RED algorithm. Section 3 will make a brief overview on the traffic conditional algorithms. In section 4, we discuss the metrics that will be used in our simulation. Section 5 describes the simulation environment and results of our simulation. In section 6 we conclude this paper.

2. Random early detection (RED)
Random early detection (RED), also known as random early discard or random early drop is an active queue management algorithm (AOQ). It is also a congestion avoidance algorithm. This algorithm plays an important role in avoiding full queues, reducing the packet delay and loss. It monitors the average queue size and dropped packets based on statistical probabilities. The RED is also known as a threshold based queuing manager. It statistically drops packets from flows before it reaches its hard limit. So it is considered a good queue for backbones, where you can't afford the complexity of per-session state tracking needed by fairness queuing. For every packet arrives, the RED gateway calculates the weighted moving average queue size (avg). It then compares the results with minimum (min_{th}) and a following schema [5,6].

- When avg < min_{th} the packet is not dropped.
- When $min_{th} \leq$ avg < max_{th} the packet is dropped with probability.
- When $max_{th} \leq$ avg the packet is dropped with probability one.

3. Traffic conditional
It is an entity which performs traffic conditioning functions. It may contain meters, markers, droppers, and shapers. Traffic meters measure the temporal properties of the stream maximum (max_{th}) threshold according to the of packets selected by a classifier against a traffic profile. The meter then passes state information to other conditioning functions to trigger particular actions. Packet markers set the DS field of a packet to a particular codepoint, adding the marked packet to a particular DS behaviour aggregate. Shapers delay some or all of the packets in a traffic stream in order to bring the stream into compliance with a traffic profile. A shaper usually has a finite-size buffer, and packets may be discarded if there is not sufficient buffer space to hold the delayed packets. Shapers delay packets in a stream in order to bring the stream into compliance with a traffic profile. Droppers discard some or all of the packets in a traffic stream in order to bring the stream into compliance with a traffic profile.

3.1. Single rate three color marker (srTCM)
The srTCM basically consists of two token buckets representing two burst sizes with both token buckets being filled by a single rate called the Committed Information Rate (CIR). The policing is done on the length and not on the peak rate of the burst. So, packets are metered and marked according to the three parameters (CIR), the Committed Burst Size (CBS) and the Excess Burst Size (EBS). A packet is marked green if it doesn't exceed the CBS, yellow if it does exceed the CBS, but not the EBS, and red otherwise [7].

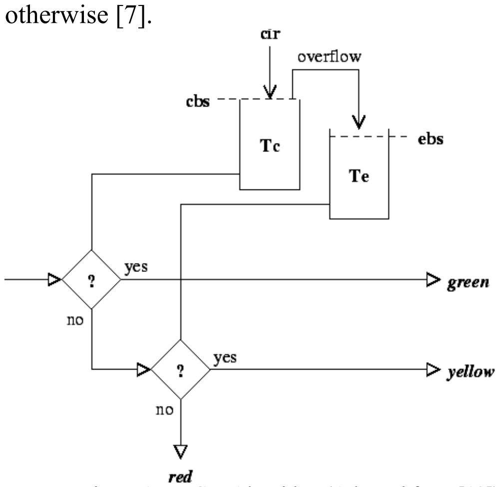

Figure 1: srTCM Algorithm (Adopted from [13]).

Figure 1 gives a complete description of the srTCM algorithm.

3.2. Two rate three color marker (trTCM)
The trTCM consists of two token buckets representing two burst sizes with the token buckets being filled by two different rates representing CIR and the Peak Information Rate (PIR). The packets are metered and marked according to the four traffic parameters – the Peak Information Rate (PIR) and its associated Peak Burst Size (PBS) and the Committed Information Rate (CIR) and its associated Committed Burst Size (CBS). The PIR must be greater than or equal to the CIR. The PBS and the CBS are both measured in bytes and must be configured to be greater than zero. It is recommended that they are

configured to be equal to or greater than the size of the largest possible IP packet in the stream [8]. A packet is marked Red if it exceeds the Peak Information Rate (PIR). Otherwise, it is marked either Yellow or Green, depending on whether or not it exceeds the Committed Information Rate (CIR) [9]. A detailed description of the trTCM algorithm is shown in figure 2.

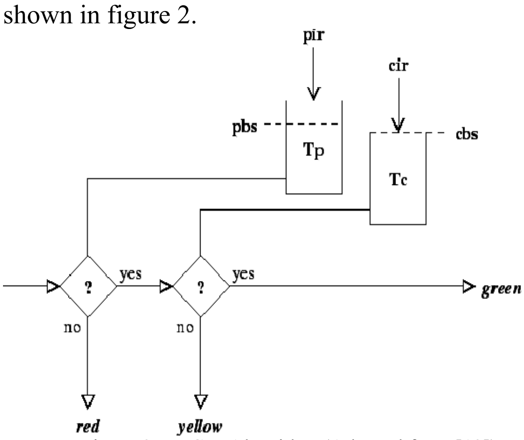

Figure 2: trTCM Algorithm (Adopted from [13]).

4. Metrics

This section will focus on the metrics and measures that will be used through in the simulation. The fairness index and the stander deviation will be the used metrics. Both metrics will be performed over the goodput of every node.

4.1 Fairness index

It is a measure used in network engineering in order to determine whether users or applications are receiving a fair share of system resources. Fairness is an important subarea of QoS. Customers will not directly see the fairness issues, but fairness may have a remarkable effect on performance of individual flows [10,11]. In our simulations we will use this measure to get the effect of the RED parameters on the fairness index of the goodput.

$$f(x) = \frac{\left[\sum_{i=1}^{n} x_i\right]^2}{n \sum_{i=1}^{n} x_i^2} \qquad (1)$$

Where n represents the total number of UDP sending nodes and x represent the goodput of every node.

4.2. Standard Deviation (SD)

The standard deviation measures the spread of the data about the mean value. It is useful in comparing sets of data which may have the same mean but a different range. Through this paper, we will use the SD as a fairness measure.

The use of the SD will allow us to significantly monitor the change in the number of goodput bits for each dropping probability. As the fairness values are between 0 and 1, the change in the fairness values are not sensitive same as the change in the SD values.

5. Simulation

In this section we will introduce our simulation environment, the used simulator and the simulation results. The results will be separated into two parts one for the packet size 1024 bits and the other for packet size 2048 bits. In each part we will find out the effect of the RED probability on the packet loss and goodput of every node.

5.1. Simulation environment

Figure 3 shows our simulation network topology which will be used. As shown in the figure, we have core router C1 and three edge routers E1, E2, E3. All routers implement the active queue management algorithm. The RED is setup to use three virtual queues for the red, yellow and green packets. The setting for the red, yellow and green packets are {0, 40, 0.2}, {40, 80, 0.1} and {80, 120, 0.02} respectively. Through out our simulations, we have two parameters that at least one of them changes in each simulation. Those parameters are the voice packet size and the packet dropping probability of the third virtual queue.

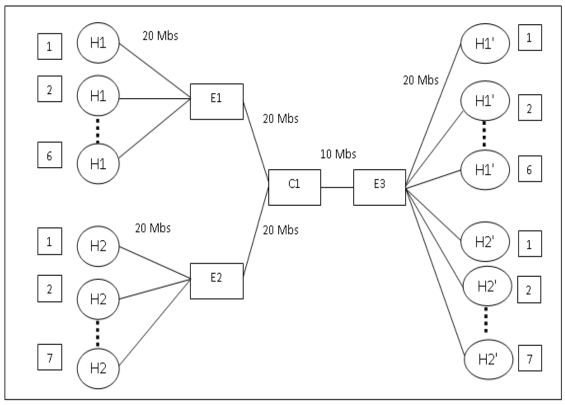

Figure 3: simulation topology.

The packet size would either be 1024 bits or 2048 bits and the dropping probability would range from 0 to 1 with a 0.1 increase. For the srTCM and trTCM, the CIR and voice rate are set to 1024 bit/s. CBS is set to 30000 bytes. For the srTCM, PBS is set to 36000 bytes. For the trTCM, PIR is set to 1200 bit/s.

5.2. Simulation results

Our simulation results will be separated into two cases. First case will be about the srTCM. The second will be about the trTCM. For each policy marker, the packet size will be either 1024 bits or 2048 bits; the drop probability of the red marked packets will vary from 0 to 1. These changes in the drop probability are done on both edge and core routers at the same time. The number of lost packets and SD and fairness index of the goodput of every node will be measured for every case.

5.2.1 Simulating the srTCM Algorithm

The following graphs explain the effects of the drop probability of the third virtual queue (red marked packets) on the srTCM when packet size is 1024 bits and 2048 bits.

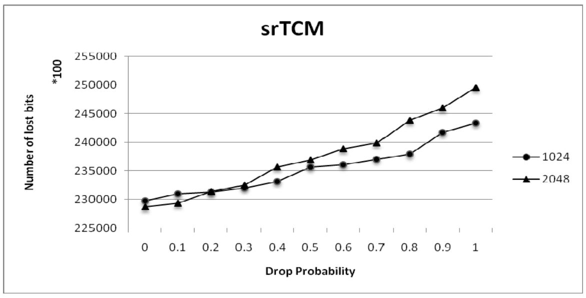

Figure 4: Number of lost bits versus the packet drop probability for both packet sizes.

Figure 4 shows the effect of dropping probability on the lost bits when packet size is either 1024 or 2048 bits. We see that the dropping probability of the third virtual queue has a great effect on the lost bits. This effect is for both packet sizes. The increase in the dropping probability leads to an increase in the lost bits for both packet sizes. As it is known, this increase leads to increase the chance of dropping the red packets from its queue. So with this increase in the dropping probability, the number of the lost packets will increase as well.

We see also the packet size has effect on the lost bits. As shown, in the first three dropping probabilities the lost bits when the packet size is 2048 are less than the lost bits when packet size is 1024 bits, but after that lost bits when the packet size is 2048 bits, are greater than the other. This happens because the queue size is configured in number of packets, not number of bits. So, the number of dropped packets will be small when the dropping is small and the queue is full for packet size 2048 bits. This means that the number of the goodput bits will be larger than the number of goodput bits when the packet size is 1024 bits.

Under the same conditions but with high dropping probability, the number of lost bits when packet size is 2048 bits will be greater than the number of lost bits when packets size is 1024 bits.

Figure 5 shows the effect of the dropping probability on the fairness index of the goodput of every VoIP node. As it is shown, the first three dropping probability values make a significant change in the fairness index. But after that, the fairness index changes very slightly up and down. The reason for this significant change in the fairness index at the first three drop probabilities is that when the third queue becomes full by the first sending nodes, the other coming packets from the other nodes will be dropped as the first, second and third queues are full. This leads to unfairness treatment between the first sending nodes and other node which may send just after.

The packet size effect is very slight. This happens because the queue size is configured in number of packets, not number of bits.

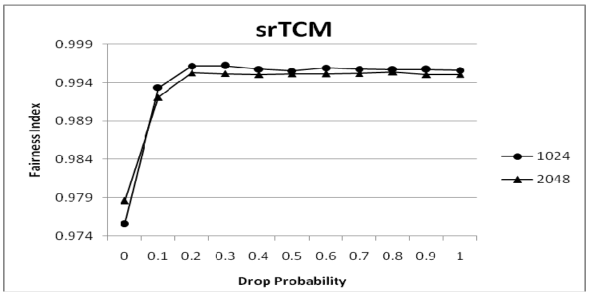

Figure 5: Fairness index versus the packet drop probability for both packet sizes

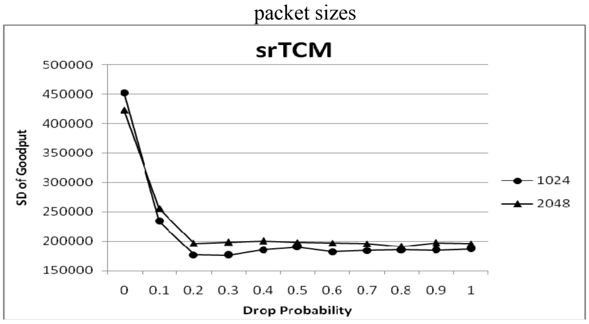

Figure 6: SD of the goodput bits versus the packet drop probability for both packet sizes. sizes

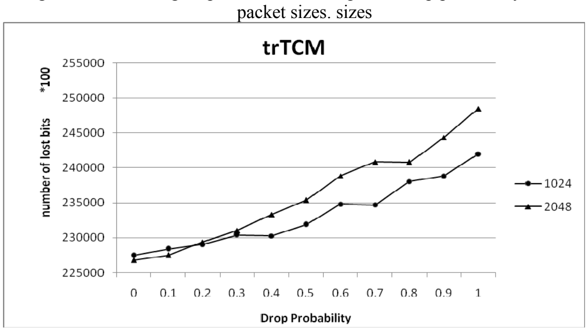

Figure 7: Number of lost bits versus the packet drop probability for both packet sizes.

Figure 6 shows the packet dropping probability effect on the SD of the goodput. We see that the dropping probability has a significant effect on the SD when its value is small. This happens because the first sending nodes reserve a place in the queue of the router. This action gives the first sending nodes a higher goodput than the other nodes so the stander division becomes higher when the dropping probabilities are small. When the dropping probability becomes higher, its effect on the SD becomes very slight.

The packet size effect on the SD cannot be neglected. The packet size 1024 bits makes a better SD for all dropping probability except when its value is less 0.1. This happens because the queue size is configured in number of packets.

5.2.2 Simulating the trTCM Algorithm

The following graphs show the effect of the dropping probability and packet size on the trTCM.

Figure 7 shows the effect of dropping probability on the lost bits when packet size is either 1024 or 2048 bits. Both the dropping probability and packet size have the same effect on the number of lost bits. But we find that the number of dropped bits when applying the trTCM algorithm is less than number of dropped bits when applying the srTCM algorithm. This happens because packets in trTCM algorithm have lower probability to be marked as red packets, because the size of the first token bucket is updated PIR times every second ,while in srTCM the first token bucket size is updated CIR times per second. As it is known the PIR corresponds to the maximum rate in which the traffic of a class can be sent or received and the CIR is the rate which is guaranteed that will always be available to the respective traffic class. From CIR and PIR definitions, we get that PIR value maximum allowable rate, while the CIR is the minimum allowable rate. So the chance that a packet is marked red or even dropped is higher in the srTCM than in trTCM. As a result, number of dropped bits in the trTCM is less than the number of dropped bits in the srTCM. The packet size has the same effect on the lost bits. This happens due to the queue size configuration.

Figure 8 shows the effect of the dropping probability on the fairness index of the goodput when applying the trTCM algoritm. From the graph we get that fairness index of goodput in trTCM is affected the same as the srTCM. The Packet size also has the same effect on trTCM like the srTCM.

From figure 9 we find that, the effect of packet size and packet dropping probability on the SD are smaller when using the trTCM algorithm rather than using the srTCM algorithm. This happens because of the virtual queues sizes of the trTCM.

6. Conclusion and future work

In this paper, we introduced the effect of red packet dropping probability and the packet size on the Fairness index and lost packets via a computer simulation. We also introduced the SD as another sensitive measuring technique that can be used instead of the fairness index measuring technique. From our simulations, we found that trTCM algorithm achieves better fairness and lower bit loss. The dropping probability of the red packets has a great effect on the packet loss. We also found that the larger the SD the smaller the fairness index and the opposite. So, we conclude that the SD is conversely proportional to the Fairness index.

Nevertheless, the complexity of the global network is very difficult to implement. So, it will be useful in our future work to simulate more complex network in order to get closer to reality. Besides that, studying the behaviour of a network with different host types and different transfer protocols would be our main focus in our future work.

References

[1] J. L. Ledford, “Cut the Cord! The Consumer's Guide to VoIP,” Thomson Course Technology, Canada, 2006.

[2] F. N. Pavlidou, A. Pitsillides, D. Sisalem, H. Schulzrinne, “Research and Trials for Reliable VoIP Applications,” Computer Networks, vol. 52, no. 13, pp 2447-2449, Elsevier, April. 2008.

[3] V. A. Reguera, F. F. Alvarez, W. Godoy, E. M. Garcia, “On The Impact of Active Queue Management on VOIP Quality of Service,” Computer Communications, vol. 31, no. 1, pp: 73-87, 2008.

[4] S. Blake, D. Black, M. Carlson, E. Davies, Z. Wang, W. Weiss, “RFC 2475 - An architecture for differentiated services,” Dec. 1999.

[5] S. Floyed, V. Jacobson, “Random early detection gateways for congestion avoidance,” Transaction on Networking, vol. 4, no. 4, pp: 397-413, ACM/IEEE, 1998.

[6] W. M. Eddy, M. Allam, “A Comparison of RED’s Byte and Packet Modes,” Computer Networks: The International Journal of Computer and Telecommunications Networking, vol. 42, no. 2, pp: 261-280, ACM, 2003.

[7] J. Heinanen, R. Guerin, “RFC 2697 – A Single Rate Three Color Marker,” Sep. 1999.

[8] J. Heinanen, R. Guerin, “RFC 2698 – A Two Rate Three Color Marker,” Sep. 1999.

[9] H. Kim, C. Yoo, W. Jung, “Simulation Study on the effect of the trTCM Parameters,” The Proceeding of 10th International Conference on Telecommunications (ICT 2003), vol. 31, no. 11, pp: 1482-1488, IEEE, 2003.

[10] H. Su, M. Atiquzzaman, “ItswTCM: a New Aggregate Marker to Improve Fairness in DiffServ,” The Proceeding of Global Telecommunications Conference (GLOBECOM 2001), vol. 3, pp: 1841–1846, IEEE, 2001.

[11] M. A. Elshaikh, M. Othman, S. Shamala, J. M. Desa , ” A New Fair Marker Algorithm for DiffServ Networks,” Computer Communications, vol. 31 , no. 14, pp 3064-3070 , Elsevier, Sep. 2008.

[12] The Network Simulator-ns-2. http://www.isi.edu/nsnam/ns/, (last visit Oct.16, 2008).

[13] Guide to IP Network Administration with Linux, http://linux-ip.net, (last visit Oct.16, 2008).

www.icgst.com

Towards a hybrid University to achieve Globalization

Kamal ELDahshan
Al Azhar University, Faculty of Sciences, Department of Mathematics
dahshan@gmail.com

Abstract
This paper presents guidelines for E-learning and virtual classroom concepts to an existing traditional university. The sub-goal is to transform it to a hybrid; virtual/classical university while the goal is to achieve student mobility in a global educational environment. In fact few universities have been and will be created from scratch to be virtual. The high impact is in existing traditional ones. For educational and social reasons, we prefer the hybrid (virtual/ traditional) solution. These reasons will not be discussed here. On the other hand, this paper focuses on the methodology and technical issues related to the above mentioned concepts to existing universities.

Keywords: *e-learning, virtual classroom, virtual university, Hybrid University, digital learning objects, FOSS.*

Introduction
In this paper we introduce the concept of a hybrid (virtual/traditional) university and show how to introduce it to already existing traditional universities. Architecture of the proposed solution is presented. The use of free and open source software is introduced and justified.
The rest of the paper is organized as follows; section one is the literature review, section two introduces the basic concepts; portal, e-learning, virtual classroom and virtual university. Then the hybrid university concept is introduced. Section three proposes Architecture for such a university. Section four applies the proposed architecture to a case study. Finally, the conclusion and further work are emphasized in section five followed by the references in section six.

1. Literature Review
Many papers were published to support the creation of a virtual university. Some of them, are at the micro level. These study, examine and develop the necessary technical infrastructure for a virtual or pseudo-virtual university.
While others, like this one are at the macro level. These discuss the issue from a strategic or global point of view, without neglecting the implementation.
Elmahdy & Fahmy [13] Proposed a low cost hardware solution for webcams and points out needs and limitations to apply webcam in local environment. The main advantages of the proposed system are portability, reliability, open source, high security, high performance. They planned to redesign the system, but using video streaming.
Mosteghanemi & Balla [10] Implemented a platform that incorporated at least one course. They used Moodle under windows. Although this is to be classified, like the previous one at the micro level, their title is ambitious towards the creation of a numeric university (another synonym of a virtual university).
Pigliapoco et al. [21] presented a chat tool called LOL classroom, which recreates the environment of a physical university classroom. This is clearly at the micro level, but it does not stop there. This chat tool is then integrated with the LOL e-learning platform (Land Of Learning by MEC informatics).
Hiltz et al. [4], explored and presented the most significant factors that motivate and inhibit faculty with regard to teaching in online environments. They found that "Leading motivations are the flexibility allowed by being able to teach anytime/anywhere; better/ more personal interaction and community building supported by the medium; the technical and creativity challenges offered by this mode of teaching; being able to reach more students; and better course management". On the other hand they found that the "major sources of dissatisfaction are more work, medium limitations, lack of adequate support and policies of teaching online, and the fact that the medium is not a good fit for some students.
In fact, we care about the faculty. Their adaptability, capabilities and satisfaction are a key to success, when we talk about the movement towards a hybrid learning university.
For a real application, the reader may visit the
Stanford University presents an ongoing experience that seems to be working [17].

2. Basic Concepts

FOSS

Free and open source software FOSS, is a concept, a way of thinking and an approach to be adopted, if the global education is targeting success, especially in developing countries. In many cases the virtual university project may be stopped for budgetary reasons. Sustainability can hardly be achieved, if the students pay less for social reasons and the infrastructure initial and maintenance cost is continuously increasing. [20] Describes the development and need for free and open source movements in software development. It then proceeds in its second part to detail the need for open source content in education. UNESCO adopted Foss for e-learning. The FOSS community of interest was transferred to UNESCO's communication and information sector, home of the UNESCO free software portal [22].

Educational Portal

This is a portal for an educational organization. It contains the necessary portlets that lead to educational services. An educational-portal is vital in the success of a virtual or hybrid university. It is now even necessary for a traditional university. An educational portal should comply with the knowledge portal functionalities (communication, collaboration, services, contents, coordination, customization, community and connections) [12].

Learning Objects

The concept of learning objects, LO's is not new. It is decades old. The Digital Learning Object (DLO) is what we care about here. It is simply a learning object on the computer. It is preferably, interactive, colorful, attractive, self explanatory, stand alone, and achieves an educational objective. Consequently a DLO preferably uses multimedia. Many DLO's are available on the web. They may be downloaded and used. In many cases they are simply called learning objects. Zuckerman presents a historical overview and classification of traditional and digital learning objects. A list of free learning objects resources may be found at University of Cincinnati library web site [14].

Distant Education is the opposite of traditional face-to-face (F2F) education. In F2F education the instructor is facing the students during the education process. i.e. they have to exist at the same place at the same time. If they are not at the same place and/or the same time during the education process, then the education is considered distant.

E-learning stands for electronic learning. This happens when electronic devices are used to partially or totally support the education process. Sometimes, it is mixed with distant education.

Virtual Classroom is a simulation of a real classroom on the web. Course content delivery, explanations, discussions, collaborative work and assessments are all accomplished through the web.

Virtual University is a university where the courses are introduced to students in virtual classrooms.

The Concept of hybrid (virtual/traditional) Solution

This concept is two fold; some courses are introduced in traditional classrooms, while others are presented in virtual classrooms. Even the courses introduced in traditional classroom are divided to completely traditional and partially traditional courses. A partially traditional course uses (partially) e-learning techniques to present material, quizzes, exams, or even to support the instructor inside the classroom. This will make use of the learning objects mentioned earlier.

A Shareable Content Object (SCO) is a launch able learning object (resource) that communicates with the run-time environment that launched it.

Learning Management System (LMS) is a term used to describe software tools assigned to manage user learning interventions.

Learning content management systems or LCMSs are systems provide tools to deliver and manage instructor online training based on learning object methodology.

The Learning Systems Technology Architecture (LTSA) is an architecture based on abstract components.

An asset: is a simple resource, such as a static HTML page or a PDF document, or collection of files, such as images and a style-sheet, which does not make use of the run-time API defined by SCORM. Therefore an asset does not communicate with the run-time environment delivering it.

3. The proposed Architecture

The proposed solution is based upon applying the concept of hybrid solution mentioned above.

3.1. Methodology

M0- The necessary hardware and software infrastructure is a preliminary step for the introduction of virtual classrooms in a traditional university. This has to be taken care of at the university level, the instructor level and the student level.

M1- A library of DLO's is to be built, constructed, downloaded and maintained.

M2- The first step is to convince traditional senior instructors to introduce e-learning in their courses. This begins with power point presentations, and then proceeds with using DLO's. They will need a lot of help to imagine where, which and how DLO's may give them support in their job. This is easier with science, medicine and engineering for example, than human studies. Multimedia and interactivity are usually more impressive when applied with the former.

The role of an existing and growing library of DLO's is essential at this stage. Students' projects may be useful to sustainably enrich this library.

Assessment can also be used to convince senior instructors. Automatically evaluated quizzes and exams have always been a dream of heavy loaded instructors. This is another entry point to senior instructors, minds to encourage them to be e-learning users in a traditional university.

M3- The next step is to allow the instructors to have some days off! Let the computer do the job for you!! We may help you prepare some course sessions using software. On the job training of senior instructors on the necessary software is a must. The choice of a good FOSS is essential at this stage. The senior instructor would not like to pay a penny of his budget for that e-learning story. The availability once again of the DLO's library would also play an important role here.

M4- The senior instructor is now, after a couple of semesters or so, ready to accept the idea to have his class divided into two groups; one partially virtually attending his course and the other completely virtually. Administrative regulations that allow the students to easily move from one group to another are essential for that success of the evolution towards the hybrid university.

M5- Junior instructors will be much easier to orient and be trained to use the virtual classroom concept.

M6- Standards play an essential role in the introduction of the hybrid system. Standards will first assure that the virtual course is equivalent to the traditional one in all aspects, educational and administrative. This is a must to facilitate students transition from one classroom to another. It is also vital to allow the students mobility from one university to another. Standards concern curricula, credit hours (per course and per degree), instruction methods (traditional, e-learning, interactive…), assessment and the instructors themselves. Examples of standards are: the ACM/IEEE curricula of computer science and information systems, Scorm for educational material and so on.

M7- Having virtual classes in more than one university at the same country or region will accompanied with suitable legislations will facilitate and encourage students mobility from one university to another, nevertheless a minimum of requirements may be imposed by each university of the student to be awarded a degree from that university. This may include without being limited to a minimum GPA, a minimum number of credit hours gained at the university and a total number of credit hours achieved by the student.

3.2. Our Approach

The approach presented in this paper is based upon the gradual introduction of the e-learning concept in a traditional university. Instructors will build their traditional courses the new way using the new toy (IT based). Then the

1- On top of the (preferably open source) operating system, a content management system CMS, accompanied with a learning management system LMS are installed; both FOSSES.
2- A collection of courses is designed and implemented using this infrastructure. Assessment should also be handled.
3- The accompanying managerial environment is installed. It should also be FOSS.
4- The environment is used to build a suitable information system for the virtual university.
5- A virtual classroom tool (FOSS) compatible with the preceding CMS and LMS is added to the system.
6- The virtual classroom tool is used to enable the LMS users to live the virtual environment, chat, audio and video communication, whiteboard and content.
7- A portal engine (FOSS) should then be added to the system and used to build an educational portal suitable for the virtual university public (staff, students …) needs.

3.3.Architecture

Thus the system architecture may be represented as follows:

4. The Case Study

4.1. Application

We implemented the approach mentioned above on four courses at different faculties and with the help of different teams of students.

1- The software infrastructure included LMS: Moodle, CMS: Reload.
2- The courses chosen are database systems (introduction to, and advanced), Web programming, ICDL preparation course and a completely theoretical course about logic (not mathematical).
   The former two are presented to computer science students.
   They are being taught and consequently implemented in
   English. While the other two are taught and consequently
   implemented in Arabic. To mention here the availability of an Arabic version of moodle, which supported our implementations easily.
   The course material, quizzes, exams, etc. have been prepared and integrated. Figures (2, 3) show quiz preparation:
3- A managerial environment included, Apache, MYSQL and php.

4- A tailored information system was built to support the virtual university using the previously mentioned environment.

5- Wiziq virtual classroom tool was chosen and installed.

6- Wiziq was used to integrate the courses built using Moodle in the virtual environment. figure(2)

7- The portal engine chosen was phpnuke. It has been used to create the convenient educational portal.

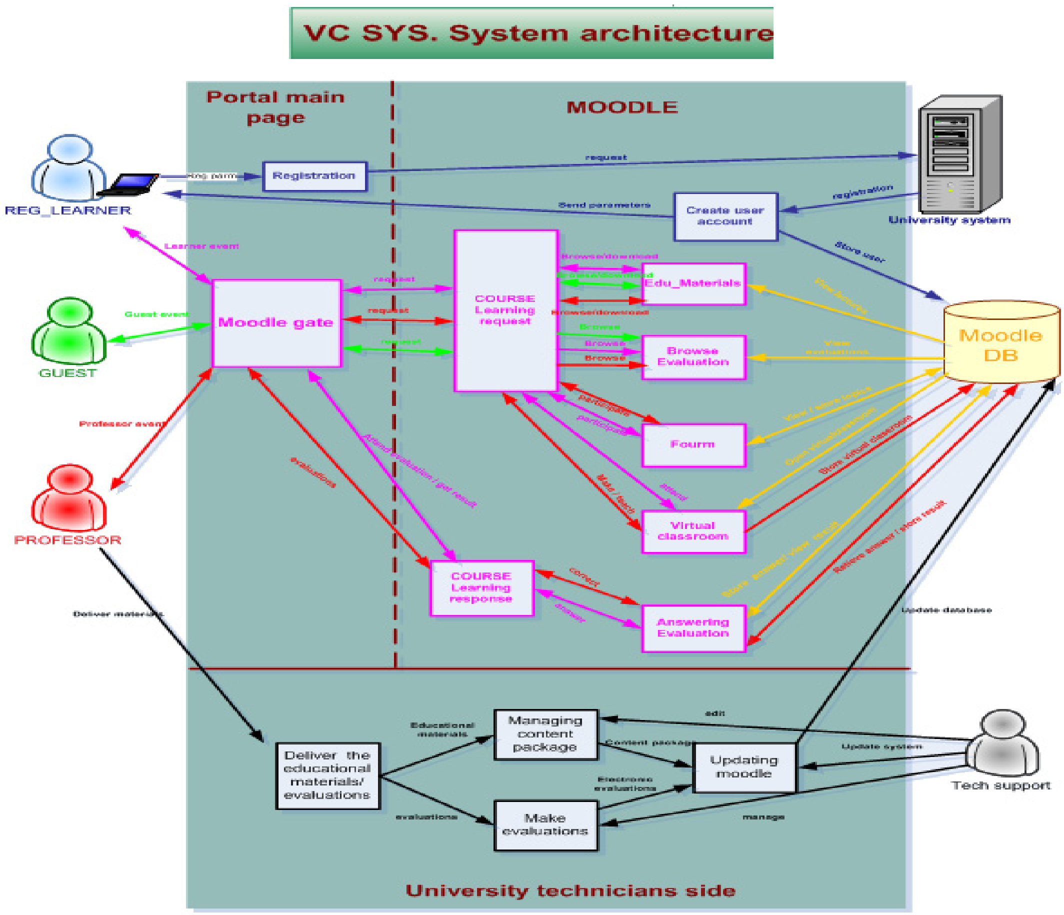

Figure1. System Architecture

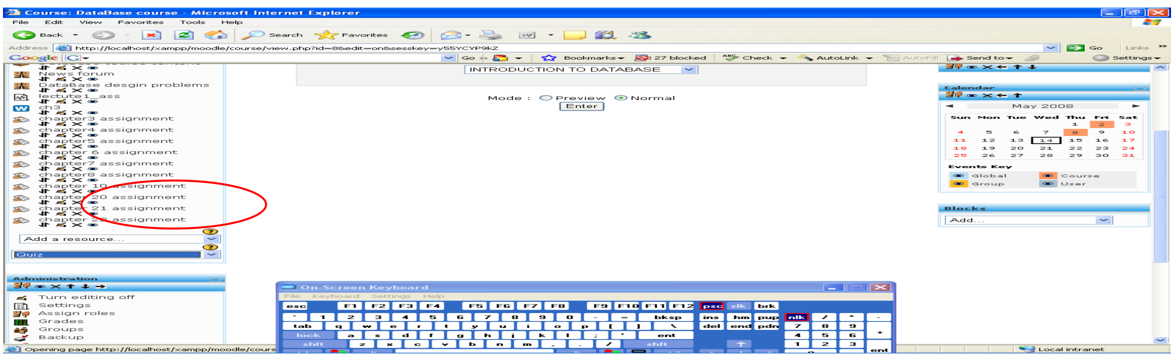

Figure 2. Quiz Active

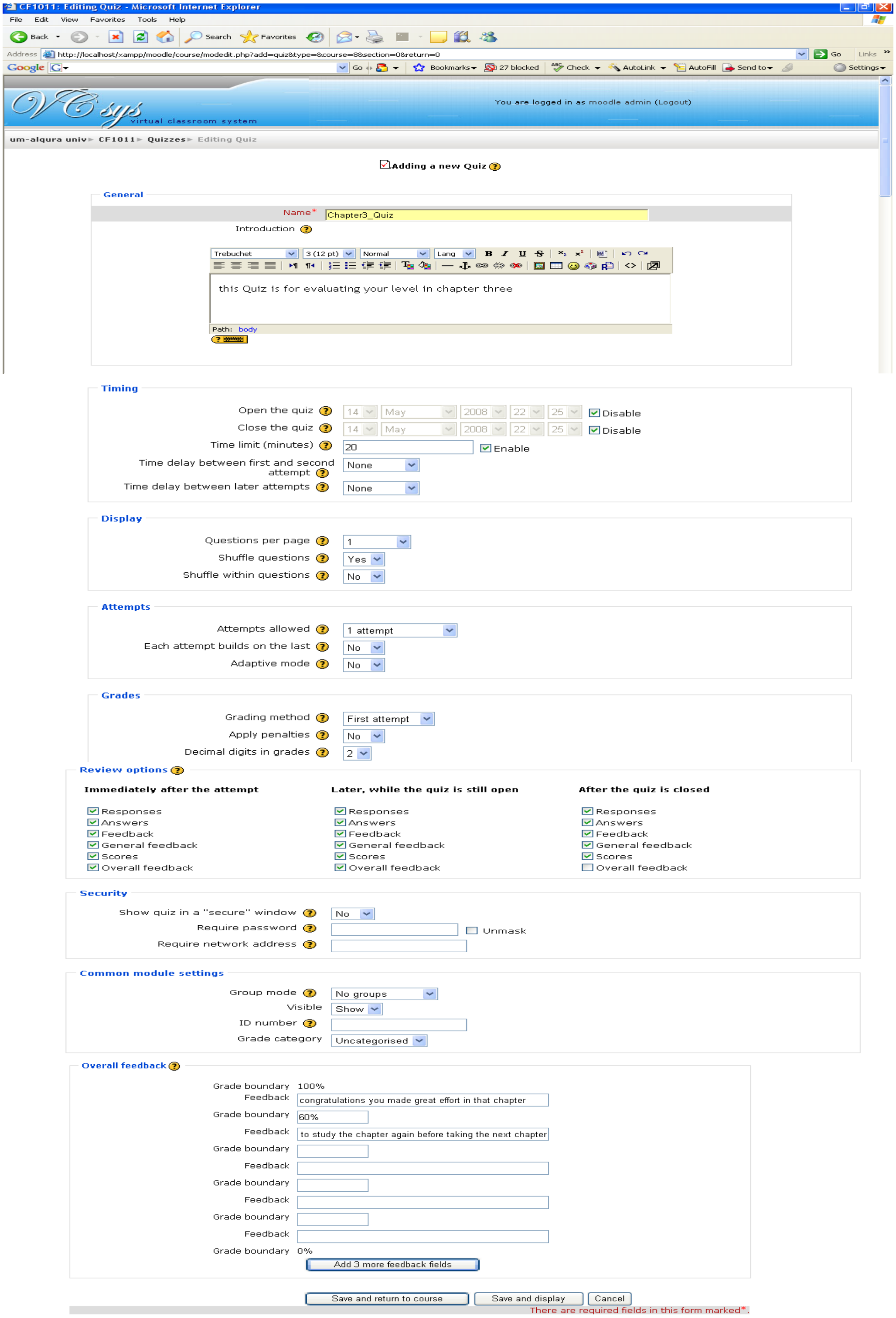

Figure 3. Quiz Prepared

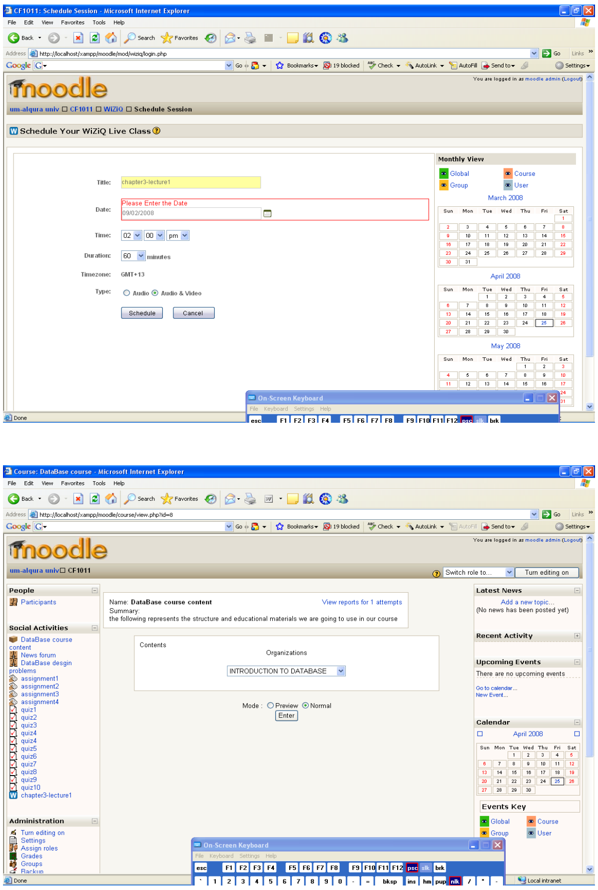

Figure 4. Adding a virtual classroom

4.2. Software Layers

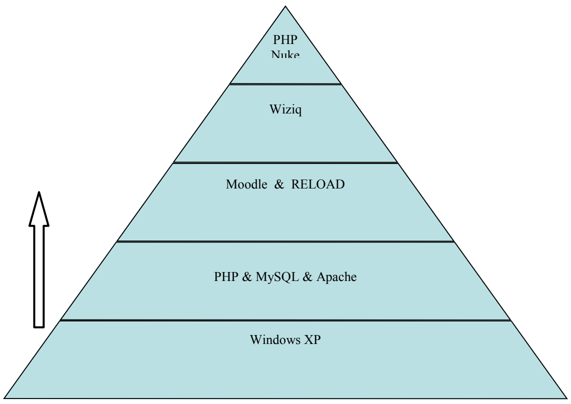

Figure 5. Software Layers

5. Conclusion and Further Work

This paper explained how to move from a traditional university to a hybrid (traditional/virtual) one. It presented the methodology, the new approach and a case study to apply the theoretical investigation.

The solution supports and is supported by FOSS.

A long term plan was presented to introduce the Virtual classroom and virtual university concepts to a traditional one. The evolution and system maintenance have also been covered.

This effort should be accompanied with adequate legislation to guarantee the students enrollment in the new system.

After success and stability in one university (pilot project), other universities in the same country, or region may follow the same approach. Accompanied with the necessary legislations, this will enable and facilitate student mobility. This leads to more freedom, better service and lead to a more productive society.

6. References

[1] Hong Joo Lee, Jong Woo Kim, Joon Koh : "A contingent approach to knowledge portal design for R&D Teams, relative importance of knowledge portal functionalities", Expert Systems with applications, Feb. 2008.

[2] Pei-Chen Sun, Hsing Kenny Cheng, Tung-Chin Lin, Feng-Sheng Wang, "A design to promote group learning in e-learning: Experiences from the field" Computers & Education Volume 50, Issue 3, PP 661-677 April 2008.

[3] Hemmendinger, D., "The ACM and IEEE-CS Guidelines for Undergraduate CS Education" Vol. 80, No.5 communication of the ACM May 2007.

[4] Starr Roxanne Hiltz, Eunhee Kim and Peter Shea Hiltz, S.R., Kim, E. and Shea, P. “Faculty Motivators and De-motivators for Teaching Online: Results of Focus Group Interviews at One University”. Proceedings of the 40th Hawaii International Conference on System Sciences, 2007.

[5] M. Notess and L. Lorenzen-Huber,_" Online Learning for Seniors: Barriers and Opportunities" E-learn magazine, ACM, 2007.

[6] S. George , H. Labas," E-learning standards as a basis for contextual forums design" Elsevier 2007.

[7] E. Melis, G. Goguadze, M. Homik, P. Libbrecht,C. Ullrich and S. Winterstein,"Semantic-aware components and services of ActiveMath", British Journal of educational Technology Vol. 37 No 3405–423, 2006.

[8] C. Romero, S. Ventura, " Data Mining in e-Learning", WIT Press, 2006.

[9] H. Hedden, “Creating an Independent Online Course for a Niche Skill" elearn magazine, ACM, 2006.

[10] Mhammed Mosteghanemi and Amar Balla: “Creating a numeric university through open source Moodle platform”, AIML 06

International conference, 13-15, Sharm El Sheikh, Egypt, June 2006.
[11] M. Rosenberg" Beyond E-Learning: Approaches and Technologies to Enhance Organizational Knowledge, Learning, and Performance" John Wiley & Sons 2006.
[12] Shackelford, R., James H. Cross, else, "Computing Curricula 2005", ACM Sep.2005.
[13] Hesham Elmahdy and Ali Fahmy: “Evaluation of WebCam Architecture challenges”ICGST-GVIP Journal, Vol.5, Issue1, December 2004.
[14] John T. Gorgone, Gordon B. Davis, Joseph S. Valacich else, "IS 2002 Model Curriculum and Guidelines for Undergraduate Degree Programs in Information Systems", ACM, 2002.
[15] Chang, C., Denning, P. J., else, "Computing Curricula 2001 Computer Science" the National Science Foundation under Grant No. 0003263, Dec.2001.
[16] Marc J. Rosenberg, "E-Learning: Strategies for Delivering Knowledge in the Digital Age", McGraw-Hill, 2001.
[17] Stanford: http://see.stanford.edu/ State of 20th of January 2009.
[18] http://www.toolbook.com Feb. 2008.
[19] www.virtualcampus.ch Feb. 2008.
[20] www.elearnspace.org Feb. 2008.
[21] Erika Pigliapoco, Giovanni Torrisi. Massimo Messina, Alessandro Bogliolo: “LOL classroom: A virtual University classroom based on enhanced chats”, EURODL, 2008.
[22] www.unesco.org

Biography

Dr. ElDahshan is an associate professor of Computer Science and Information Systems at Al-Azhar University in Cairo, Egypt. An Egyptian national and a graduate of Cairo University, he has obtained his doctoral degree from the Université de Technologie de Compiègne in France, where he has also taught for several years. During his extended stay in France, he has also worked at the prestigious Institut National de Télécommunications, in Paris.

Dr. ElDahshan's extensive international research, teaching and consulting experience has spanned four continents and includes academic institutions as well as government and private organisations. He has worked as a visiting professor at Virginia Tech, VA, USA, as well as a Consultant to the Egyptian Cabinet Information and Decision Support Centre (IDSC), and a senior advisor to the Ministry of Education and Deputy Director of the national Technology Development Centre. Recently he organized AlAzhar conference on computers and their applications.

Dr. ElDahshan is currently a visiting professor at Umm-al-Qura University in Saudi Arabia.

www.icgst.com

Simulation Analysis of QoS parameters by combining MAC and TCP in MANETS

J.PREMALATHA* AND P.BALASUBRAMANIE**

** ASST. PROFESSOR, DEPARTMENT OF COMPUTER SCIENCE AND ENGINEERING*

KONGU ENGINEERING COLLEGE, ERODE, TAMILNADU, INDIA

E-mail: premalatha1969@yahoo.com

***PROFESSOR, DEPARTMENT OF COMPUTER SCIENCE AND ENGINEERING*

KONGU ENGINEERING COLLEGE, ERODE, TAMILNADU, INDIA

E-mail: pbalu_20032001@yahoo.co.in

Abstract

This paper evaluates the performance evaluation of interaction between Transport and the MAC layer protocols operating in a mobile adhoc network. In Adhoc networks, certain QoS parameters like error rate, delay and packet loss are increased and certain parameters like throughput and delivery ratio are decreased in Transport layer is due to MAC problems and disconnection is also possible due to mobility or power failure. So, combine the mechanisms of these two layers to improve the QoS drastically so that people can design the network based on their requirements. We examine the effects of two different MAC protocols— IEEE 802.11and IEEE802.11e with Slow start and Arithmetic Increase and Multiplicative Decrease (AIMD) mechanism of TCP.IEEE802.11uses distributed coordination function (DCF) where IEEE802.11e uses enhanced distributed coordination function (EDCF). Specifically, we access the impact of multiple wireless hops and node mobility on the throughput performance of TCP on each MAC protocol. Additionally the other Qos parameters of throughput, delay, Bandwidth delay product, delivery ratio and packet loss using slow start and AIMD of TCP mechanism with two different MAC protocols is also investigated. Results show that in all instances, the QoS parameters 15-20% improvement in throughput, 40-45% improvement in bandwidth-delay product, 10-15% improvement in delivery ratio, packet loss is reduced drastically to 40-50% in IEEE802.11e with slow start than IEEE802.11. Results shows, the QoS parameters 35-40% improvement in throughput, 25-30% improvement in bandwidth-delay product, 15-20% improvement in delivery ratio, packet loss is reduced drastically to 20-25% in IEEE802.11e with AIMD than IEEE802.11with AIMD. Further analysis show that IEEE802.11e (EDCF) with slow start better than IEEE802.11e (EDCF) with AIMD and delay is 10msec in IEEE802.11e with slow start and it is increased to 35msec in IEEE802.11e with AIMD.

Keywords: *Mobile adhoc networks, Medium access control (MAC), Transport layer Protocol (TCP), Slow start, AIMD and Quality of Service (QoS)*

1. Introduction

In the near future, researchers envision a truly ubiquitous computing environment that will allow users to communicate from anywhere and at anytime. Mobile adhoc networks (MANETs) are part of this vision and aim to provide communication capabilities to areas where limited or no communication infrastructure exist; or, where it is simply more convenient to allow the communication devices to form a dynamic and temporary network among themselves. A "mobile adhoc network" (MANET) is an autonomous system of mobile routers (and associated hosts) connected by wireless links. In current wireless networks, WIMAX or WIFI the wireless mobile node is never more than one hop from a base station that can route data across the communication infrastructure. However, in mobile adhoc networks, there are no base stations. Instead, routing functionality is incorporated into each mobile host and, because of a limited transmission range; multiple hops may be required to allow one node to communicate with another across the ad hoc network[1]. The routers are free to move randomly and organize themselves arbitrarily; thus, the network's wireless topology may change rapidly and unpredictably. Thus, MANETs can be characterized as having a dynamic, multi hop, and constantly changing topology. While mobile adhoc networks can be used in a standalone mode—where there is no fixed infrastructure, their use is also being considered as an extension to the Internet.

Thus, the effectiveness of the wireless medium access control (MAC) protocol and mechanisms will play a central role in the success of MANETS. Several MAC protocols have been developed for wireless environments (i.e. wireless LANS) such as Carrier Sense Multiple Access (CSMA), Multiple Access with Collision Avoidance (MACA), Floor Acquisition Multiple Access (FAMA), IEEE802.11 and IEEE 802.11e. Each MAC protocol is based on multiple design choices and utilizes distinct medium access mechanisms.

This research centers on investigating the performance of and interaction between TCP and two different MAC protocols— IEEE 802.11and IEEE 802.11e, operating in mobile adhoc networks. Reliable data transfer and congestion control are key requirements for any computer network. TCP, which fulfills both of these requirements, is the most widely used reliable transport protocol in today's Internet and has demonstrated its viability with respect to Internet connectivity. TCP is used to transport a significant portion of Internet traffic such as e-mail (SMTP), file transfers (FTP), and WWW (HTTP). Thus, the use of TCP in mobile adhoc networks is clearly advantageous. However, the defining characteristics (e.g., time-varying, dynamic, multihop, and constantly changing topology) of mobile adhoc networks may result in unpredictable link failures resulting in the poor performance of TCP.

The goal of this paper is, therefore, to study the effects of these characteristics on the performance of and interaction between TCP and the MAC layer protocol operating in a mobile ad hoc network. This includes examining the effects of IEEE 802.11 and IEEE 802.11e MAC protocols on the performance of TCP. Specifically, we access the Qos parameters throughput, delay, Bandwidth delay product, and delivery ratio and packet loss performance of TCP as function of node mobility.

The remainder of the paper is organized as follows: Section (2) focuses on related work; Section (3) emphasizes simulation and methodology, section (4) explains the results and section (5) is the conclusion.

2. Related work

TCP has been shown to have poor performance over wireless links. Thus, several studies have focused on improving TCP performance in the wireless mobile Environment. These include end-to-end mechanisms such as TCP-SACK and ELN and link-layer protocols such as AIRMAIL, Indirect-TCP, and TCP-Snoop. Such mechanisms and protocols were designed to work in the context of cellular-based networks fixed infrastructure networks. However, the aforementioned schemes have not considered the unique characteristics of adhoc networks, namely multi-hop routing and the lack of a centralized controller and manager (e.g., base stations). Recent work has begun to evaluate the performance of TCP in context of adhoc networks. This work demonstrated the use of combining the mechanisms of both TCP and MAC protocols improve the QoS parameters. Previous work investigates that IEEE802.11e better than IEEE802.11 [2] but not combined with TCP mechanisms. Hence, evaluating the performance of TCP in a mobile adhoc environment and quantifying the effects of the unique characteristics is an open and interesting problem. These results should facilitate the development of mechanisms for improving TCP performance in adhoc networks as well as the design of efficient and scalable quality-of-service (QoS) schemes.

3. Simulation and Methodology

This simulation study was conducted using NS2 to simulate adhoc network, which consist of 60 wireless/mobile nodes roaming in a 2600 x 400m area. In this dynamic topology, the radio transmission range of each node is approximately 250 meters and the channel capacity is 2Mbits/sec. The free space propagation model is used to determine if a node is reachable. This model predicts received signal strength when the transmitter and receiver have a clear, unobstructed line-of-sight path between them. Received power decays as a function of the T-R separation distance. This study investigates the performance slow start mechanism of TCP over two different MAC protocols: IEEE802.11 and IEEE 802.11e. Both protocols requires carrier sensing before transmission and operates as follows

3.1 IEEE802.11 MAC Protocol

The basic access mechanism for both MAC protocols is the Distributed Coordination Function (DCF). DCF is essentially a Carrier Sense Multiple Access (CSMA) that incorporates Collision Avoidance (CSMA/CA) and a positive acknowledge (ACK) scheme. Receipt of an ACK (from the receiving node) indicates that no collision occurred. If the sending node does not receive an ACK, then it will retransmit the fragment until it gets acknowledged or discarded after a specified number of retransmissions. Optionally, a mobile node can utilize the virtual carrier sense mechanism, which utilizes request-to-send (RTS) and clear-to-send (CTS) exchanges for channel reservation. Using virtual carrier sensing reduces the probability of two nodes transmitting simultaneously (collisions) because they cannot hear each other (i.e. hidden terminal problem). The difference between IEEE802.11 and IEEE 802.11e is, to assign priority for user packets in IEEE 802.11e and there is no priority assignment for user packets in IEEE 802.11.In IEEE802.11 also uses exponential backoff algorithm. A station that senses the channel busy will defer its access until the channel is later sensed idle. Once the channel is sensed idle for an amount equal to DIFS, the station then computes an additional backoff time and counts down this time as the channel is sensed idle. When the random backoff timer reaches zero, the station transmits its frame [5].

The 802.11 MAC works with a single first-in-first-out (FIFO) transmission queue. The IEEE 802.11 constitutes a distributed MAC based on a local assessment of the channel status, i.e., whether the channel is busy (i.e., a station is transmitting a frame) or idle (i.e., no transmission). Basically, the IEEE 802.11 of DCF works as follows: When a frame arrives at the head of the transmission queue, if the channel is busy, the MAC waits until the medium becomes idle, then defers for an extra time interval, called the DCF Inter frame Space (DIFS). If the channel stays idle during the DIFS difference, the MAC then starts the backoff process by selecting a random backoff counter. For each slot time interval, during which the medium stays idle, the random backof is decremented. When the backof reaches zero, the frame is transmitted [8] [9].
On the other hand, when a frame arrives at the head of the queue, if the MAC is in either the DIFS difference or the random backoff process, the processes described above are applied again. That is, the frame is transmitted only when the random backoff has finished successfully. When a frame arrives at an empty queue with no ongoing backoff process and the medium has been idle longer than the DIFS time interval, the frame is transmitted immediately. Each station maintains a contention window (CW), which is used to select the random backoff counter. The backoff counter is determined as a random integer drawn from a uniform distribution over the interval [0,CW]. How to determine the CW value is further detailed below. If the channel becomes busy during a backoff process, the backoff is suspended. When the channel becomes idle again, and stays idle for an extra DIFS interval, the backoff process resumes with the suspended BC value.

The CW size is initially assigned CWmin, and increases when a transmission fails, i.e., the transmitted data frame has not been acknowledged. After any unsuccessful transmission attempt, another backoff is performed using a new CW value updated by 2 • *{CW* + 1) - 1, with an upper bound of CWmax [6]. This reduces the collision probability in case there are multiple stations attempting to access the channel. After each successful transmission, the CW value is reset to CWmin, and the station that completed the transmission performs DIFS deference and a random backoff even if there is no other pending frame in the queue. This is often referred to as "post" backoff, as this backoff is done after, not before, a transmission.

3.2 IEEE802.11e MAC Enhanced DCF (EDCF)

The DCF is supposed to provide a channel access with equal probabilities to all stations contending for the channel access in a distributed manner. However, equal access probabilities are not desirable among stations with different priority frames [3]. The emerging EDCF is designed to provide differentiated, distributed channel accesses for frames with 8 different priorities (from 0 to 7) by enhancing the DCF as shown in Table1.

Table1: EDCF user priority table

User priority	Access category	Designation
0	0	Best effort
1	0	Best effort
2	0	Best effort
3	1	Video probe
4	2	Voice
5	2	Voice
6	3	Video
7	3	Video

As distinct from the legacy DCF, the EDCF is not a separate coordination function. Rather, it is a part of a single coordination function, called the Hybrid Coordination Function (HCF), of the 802.l1 MAC. The HCF combines the aspects of both DCF and PCF.The EDCF adopts eight different priorities that are further mapped into four access categories (ACs) as shown in figure1. ACs are achieved by differentiating the arbitration inter frame space (AIFS), the initial window size and the maximum window size [7].

Four transmission queues are implemented in a station, and each queue supports one AC class, behaving roughly as a single DCF entity in the original IEEE 802.11 MAC.For the AC i (i = 0, 1, 2,3), the initial backoff window size is CWmin[i]= (Wi,0), the maximum backoff window size is CWmax[i] and the AIFS is AIFS[i]. For 0 = i <j = 3,

CWmin[i]=CWmin[j],
CWmax[i] = CWmax[j], and

AIFS[i] = AIFS[j],

and at least one of the above inequalities must be strict. If one class has a smaller AIFS or CWmin or CWmax, the class's traffic has a better chance to access the wireless medium earlier. Four transmission queues are implemented in a station and each queue supports one AC class, behaving roughly as a single DCF entity in the original IEEE 802.11 MAC.

It is assumed that a payload from a higher layer is labeled with a priority value, and it is pushed into the corresponding queue with the same priority value. Each queue acts as an independent MAC entity and performs the same DCF function with a different inter frame space (AIFS[*i*]), a different initial window size (CWmin[*i*]), and a different maximum window size (CWmax[*i*]). Each queue has its own backoff counter (BO[*i*]) that acts independently the same way as the original DCF. backoff counter. If there is more than one queue finishing the backoff at the same time, the highest priority frame is chosen to transmit by the virtual collision handler. Other lower priority frames whose backoff counters also reach zeros will increase their backoff counters with CWmin[*i*] (*i*= 0, 1.. . 3), accordingly. Use EDCF (enhanced distributed co ordination function) and Slow start and AIMD mechanisms of Transport layer enhance the MAC performance and also transport layer performance.

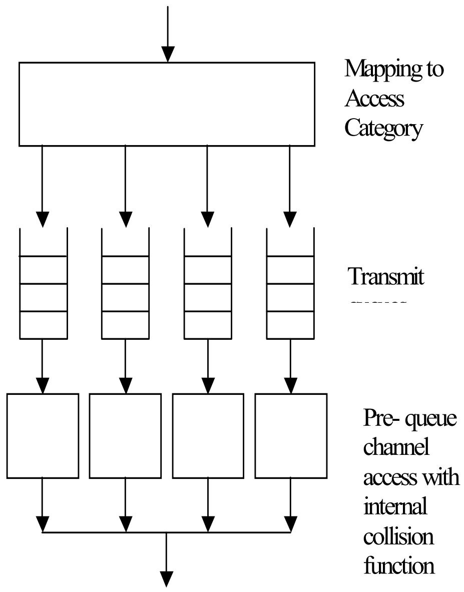

Figure1: Reference Implementation model of IEEE 802.11e

3.3 AIMD mechanisms of TCP

TCP maintain a new state variable for each connection, called congestion window, which is used by the source to limit how much data it is allowed to have in transmit at a given time. The congestion window is congestion controls counterpart to flow control advertised window. TCP is modified such that the maximum number of bytes of unacknowledged data allowed is now the minimum of the congestion window and the advertised window. TCP's effective window is revised as,

Max window = MIN (congestion window, advertised window)

Effective window = max window- (last byte sent- last byte Acknowledged)

That is, max window replaces advertised window in the calculation of effective window. Thus, a TCP source is allowed to send no faster than the slowest component the network or the destination host can accommodate. The problem, of course, is how TCP comes to learn an appropriate value for congestion window. Unlike the advertised window, which is sent by the receiving side of the congestion, there is no one to send a suitable congestion window to the sending side of TCP. The answer is that the TCP source sets the congestion window based on the level of congestion perceives to exit in the network. This involves decreasing the congestion window when the level of congestion goes down and the mechanism is commonly called additive increase/multiplicative decrease. The source determines if the network is congested and that it should decrease the congestion window based on the observation that the packets are not delivered and a timeout results, is that a packet was dropped due to congestion. It is rare that a packet is dropped because of an error during transmission.

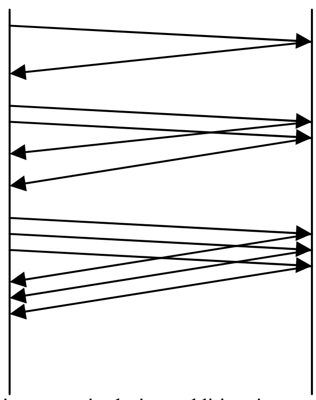

Figure2: Packets in transmit during additive increase, with one packet being added each RTT.

Therefore, TCP interprets timeouts as a sign of congestion and reduces the rate at which it is transmitting. Specifically, each time a timeout occurs, the source sets congestion window for each timeout corresponds to the "multiplicative decrease" part of AIMD.

Although congestion window is defined in terms of bytes, it is easiest to understand multiplicative decrease if we think in terms of whole packets. For example, suppose the congestion window is currently set to 16 packets. If a loss is detected, congestion window is set to 8. Normally, a loss is detected when a timeout occurs. Additional losses cause congestion window to be reduced to 4, then 2, and finally to 1 packet. A congestion control strategy that only decrease the window size is obviously too conservative. We also need to be able to increase the congestion window to take advantage of newly available capacity in the network. This is the "additive increase" part of AIMD, and it works as follows. Every time the source successfully sends a congestion window worth of packets- that is, each packet sent out during the last RTT has been Acknowledged- it adds the equivalent of one packet to congestion window. This linear increase is illustrated in Figure 2. TCP does not wait for an entire window's worth of ACKs to add one packet's worth to the congestion window, but instead increments congestion window by a little for each ACK that arrives. The order of packet transmission is x+2x+3x+4x+… where x=1

3.4 SLOW START mechanism of TCP

TCP's reaction to a missing acknowledgement is quit drastic, but necessary to get rid of congestion fast enough. The behavior of TCP shows after the detection of congestion is called slow start. The sender always calculates a congestion window for a receiver. Start size of the congestion window is one segment (TCP packet). Now the sender sends one packet and waits for acknowledgement. If this acknowledgement arrives, the sender increases the congestion window one by one, now sending two packets (congestion window= 2). After arrival of the two corresponding acknowledgements, the sender again adds 2 to the congestion window; one for each of the acknowledgements .Now the congestion window equals 4. This scheme doubles the congestion window every time the acknowledgements come back, which takes one round trip time (RTT). This is called the exponential growth of the congestion window in slow start mechanism. It is too dangerous to double the congestion window each time because the step might become too large. Therefore, the exponential growth stops at the congestion threshold .As soon as the congestion window reaches the congestion threshold, further increase of the transmission rate is only linear by adding 1 to the congestion window each time the acknowledgements come back. The order of packet transmission is $x^0+x^1+x^2+x^3+x^4$… where x=2,3,4….

4. Simulation results

The layered diagram for simulation is as shown in the figure4. To create a MANET with a collection of 60 nodes over a common wireless medium and exchanging different bytes of (160 bytes,810bytes and 1310 bytes)with different priorities of , FIFO queuing , AODV algorithm for routing . Each node is equipped with a transmitter, a receiver and a buffer used for storing data and assume that a node cannot transmit and receive at the same time (i.e., communication is half duplex). For propagation free space model has taken, it is assumed that there is only one clear line-of-sight path between the sender and the receiver.

The AODV (Adhoc On Demand Vector) protocol, available in NS2 uses dynamic routing in order to deliver packets to any destination in a mobile adhoc network and it is implemented in network layer. EDCF and DCF mechanisms are implemented in MAC layer and slow start and AIMD are implemented in Transport layer. To increase the performance there should be different types of priority level or traffic categories (TC) for data transmission in MAC layer and use user priority level of 0,1 and 2.For simulation produce 3 different packets of data and set priority 0(high priority) for large size packet, priority 1 (medium priority) for medium size packet and priority 2(low priority) for small size of packet in application layer. To send acknowledgements from transport layer in SIFS interval, a acknowledgement packet which contain less bytes of data is transmitted for all different types of traffic categories. The time slots for various inter frame spacing is set as SIFS=16µs, PIFS=25µs, DIFS=34µs, $AIFS_1$ (priority level=0 or TC1) >=34µs and every contention slot is equal to 9µs interval. If there is no high priority packet for the specified time interval immediately medium level packet are transmitted.

4.1 Results & Performance Metrics

To analyze the performance and interaction of TCP and MAC layer protocols, we evaluate them using the following metrics:

4.1.1 Throughput: It is the rate of successful message delivery over a communication channel [4]. This data may be delivered over a physical or a wireless channel and it is usually measured in bits per second (bit/s or bps), and sometimes in data packets.With 20nodes 802.11with slow start transmitted 8406 bits, 802.11e with slow start transmitted 9234 bits successfully. With same 20 nodes 802.11 with AIMD successfully transmitted 6640 bits, 802.11e transmitted 9009 bits. The Slow start mechanism of TCP with IEEE802.11e improves throughput 15-20% than IEEE802.11 with Slow start. The AIMD mechanism of TCP with IEEE802.11e improves throughput 35-40% than IEEE802.11 with AIMD. Figure5 shows comparison of Throughput performance of IEEE802.11 with Slow start, IEEE802.11e with Slow start, IEEE802.11 with AIMD and IEEE802.11e with AIMD.

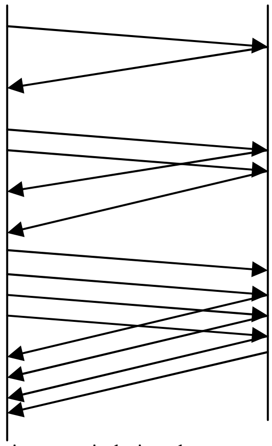

Figure3: Packets in transmit during slow start, with exponential increase in packet being added each RTT

Application Layer (This layer generate multimedia packets and assign priority)
Transport layer (Implement Slow start & AIMD mechanism)
Network layer (Implement AODV algorithm)
MAC layer (Implement EDCF&DCF algorithm)

Figure4: Layered structure

4.1.2 Bandwidth-Delay Product: It refers to the product of a data link's capacity (in bits per second) and its end-to-end delay (in seconds). The result, an amount of data measured in bits (or bytes), is equivalent to the amount of data "on the air" at any given time, i.e. data that have been transmitted but not yet received. This product is particularly important for protocols such as TCP that guarantee reliable delivery, as it describes the amount of yet-unacknowledged data that the sender has to duplicate in a buffer memory in case the receiver requires it to re-transmit a garbled or lost packet.With 20nodes 802.11 transmitted 214187.73 bits where 802.11e transmitted 240017.87 bits successfully. With same 20 nodes 802.11 with AIMD successfully transmitted 175340.95 bits, 802.11e transmitted 222086.24 bits. With 60nodes 802.11 successfully transmitted 172308.41 bits where 802.11e transmitted 244467.13 bits. With 60 nodes 802.11 with AIMD successfully transmitted 176251.11 bits, 802.11e transmitted 238604.65 bits The Slow start mechanism of TCP with IEEE802.11e drastically improves Bandwidth Delay Product 40-45% than IEEE802.11 with Slow start. The AIMD mechanism of TCP with IEEE802.11e improves Bandwidth Delay Product 25-30% than IEEE802.11 with AIMD. Figure6 shows comparison of Bandwidth Delay Product performance of IEEE802.11 with Slow start, IEEE802.11e with Slow start, IEEE802.11 with AIMD and IEEE802.11e with AIMD.

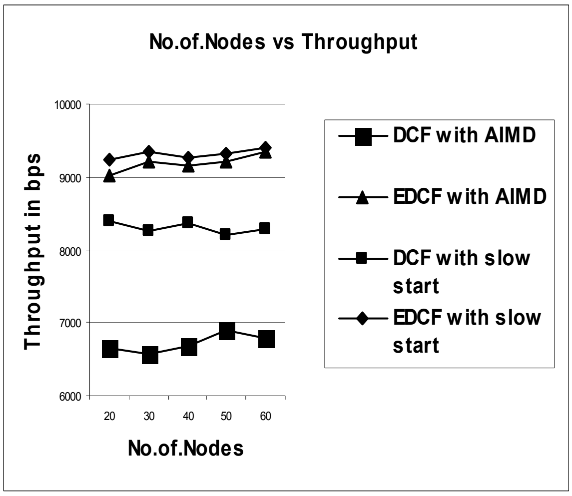

Figure5: No.of Nodes vs Throughput

4.1.3 Packet Delivery Ratio: It is the ratio between total numbers of packets received to the total number of packets transmitted. With 20nodes 802.11 with slow start transmitted 290 packets and 802.11e with slow start transmitted 317 packets successfully. With 20nodes 802.11 with AIMD transmitted 255 packets and 802.11e with AIMD transmitted 307 packets successfully. With 50nodes 802.11 with slow start transmitted 287packets where 802.11e with slow start transmitted 312 packets where as 802.11 with AIMD transmit 250 packets and 802.11e with AIMD transmit 307 packets. The Slow start mechanism of TCP with IEEE802.11e improves packet delivery ratio10-15% than IEEE802.11 with Slow start. The AIMD mechanism of TCP with IEEE 802.11e improves 15-20% than IEEE 802.11. Figure7 shows comparison of Packet Delivery ratio performance of IEEE802.11 with Slow start, IEEE802.11e with Slow start, IEEE802.11 with AIMD and IEEE802.11e with AIMD.

4.1.4 Delay: The time taken by the packets to reach the destination successfully. With 20nodes 802.11with slow start transmitted with a delay of 13msec, where 802.11e with slow start transmitted with a delay of 14msec. With 20nodes 802.11with AIMD transmitted with a delay of 11msec, where 802.11e with slow start transmitted with a delay of 15msec.The slow start mechanism of TCP with IEEE802.11e is only 0-5% higher than IEEE802.11 with Slow start and AIMD mechanism of TCP with IEEE802.11e is 35% higher than IEEE802.11. Figure8 shows comparison of Delay performance of IEEE802.11

with Slow start, IEEE802.11e with Slow start, IEEE802.11 with AIMD and IEEE802.11e with AIMD.

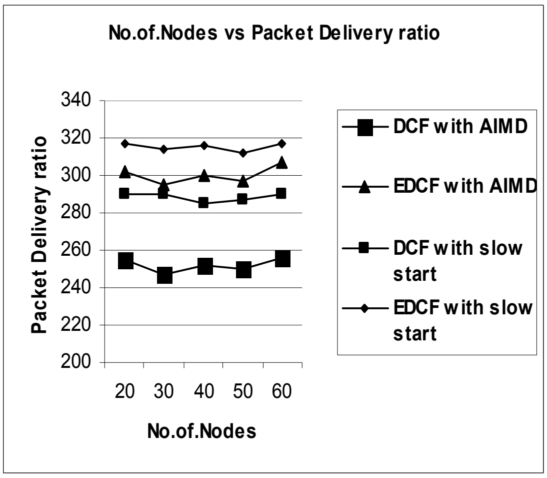

Figure6: No.of Nodes vs. Bandwidth delay product

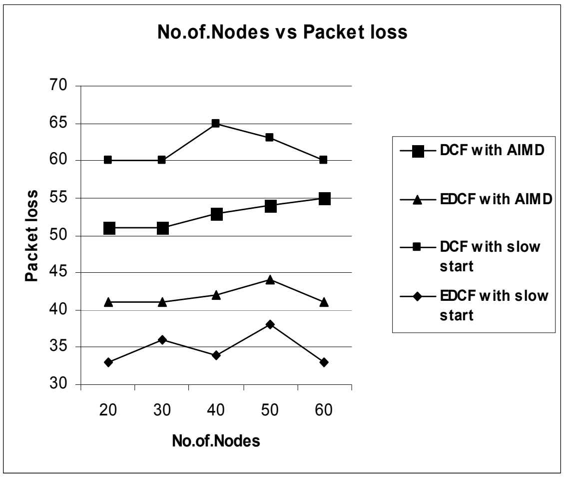

Figure7:No.of Nodes vs. Packet Delivery Ratio

4.1.5 Packet loss: The number of packets missed to reach the destination. With 20nodes 802.11with slow start missed 60packets and 802.11e with slow start missed 33 packets. With the same 20nodes 802.11 with AIMD missed 51 packets where 802.11e with AIMD missed 41 packets. The Slow start mechanism of TCP with IEEE802.11e reduces drastically the packet loss from 40-45% than IEEE802.11 with Slow start and The AIMD mechanism of TCP with IEEE802.11e reduces the packet loss to 20% than IEEE802.11.Figure9 shows comparison of Packet loss performance of IEEE802.11 with Slow start, IEEE802.11e with Slow start, IEEE802.11 with AIMD and IEEE802.11e with AIMD

5. CONCLUSION

In this paper, evaluate the performance of QoS parameters in MAC layer and its interaction with the transportation layer protocol in a mobile ad hoc network is tabulated in Table.2. This system using IEEE 802.11e and IEEE802.11 MAC mechanisms are contention based channel access function or distributed coordination function that improves quality of service in MAC layer. To improve the performance of at the transport layer will require the design of distributed medium access control scheme and proper packet transmission mechanism like slow start. A suitable MAC layer protocol and slow start algorithm improves quality of service in transport layer.

This results show that the interaction between transport layer and the MAC protocol has a significant impact on the achievable throughput, Packet Delivery Ratio, Bandwidth Delay Product and packet loss in ad hoc networks and suggest that improving scalability will result in the greatest improvement in network throughput, Packet Delivery Ratio, Bandwidth Delay Product and packet loss with minimum increase in delay. Results show that in all instances, the QoS parameters 15-20% improvement in throughput, 40-45% improvement in bandwidth-delay product, 10-15% improvement in delivery ratio, packet loss is reduced drastically to 40-50% in IEEE802.11e with slow start than IEEE802.11. Results shows, the QoS parameters 35-40% improvement in throughput, 25-30% improvement in bandwidth-delay product, 15-20% improvement in delivery ratio, packet loss is reduced drastically to 20-25% in IEEE802.11e with AIMD than IEEE802.11with AIMD. Further analysis show that IEEE802.11e (EDCF) with slow start better than IEEE802.11e (EDCF) with AIMD and delay is 10msec in IEEE802.11e with slow start and it is increased to 35msec in IEEE802.11e with AIMD.The future work is to evaluate the performance of same QoS attributes with other TCP mechanism like Arithmetic Increase and Multiplicative Decrease (AIMD) algorithm and select Dynamic Source Routing in Network Layer. Such a performance evaluation would be helpful in designing adhoc networks to satisfy user needs with required QoS parameters at both the transport and MAC layers.

6. References

[1] Altyeb Altyeb (2006),Effects of CSMA/CD on Self-Similarity of Network Traffic , ICGST-CNIR Journal, Volume (5), Issue (2), pp 7-10.

[2] Choi S (2003), 'IEEE 802.11e contention based channel access (EDCF) performance evaluation' Proc. IEEE ICC.

[3] Maarten Hoeben and Menzo Wentink, "Enhanced QoS through Virtual DCF," EEE 802.11-00/3 51,October 2000.

[4] Paolo Giacomazzi(2006), 'Quality of service for packet telephony over Mobile Ad Hoc Networks'

[5] Qixiang Pang, Soung C.Liew (2005), 'Design of an effective loss distinguishable MAC protocol for 802.11 WLAN'.

[6] Schiller J (2003), 'Mobile communication', Pearson education private limited, Singapore.
[7] Maarten Hoeben Stefan Mangold, Sai Shankar, and Menzo Wentink,"Multiple Frame Exchanges during EDCF TXOP," IEEE 802.11-01/566r3, Jan 2002.
[8] Xiao Y(2005), 'Performance analysis of priority schemes for IEEE 802.11 and IEEE 802.1 1e Wireless LANs', IEEE Transaction on wireless communication Vol 4, pp 1506-1515.
[9] Yang Xiao,(2004), 'IEEE 802.11e.QoS Provisioning at the MAC layer' IEEE wireless communications,1536- 1284, pp. 72-79.

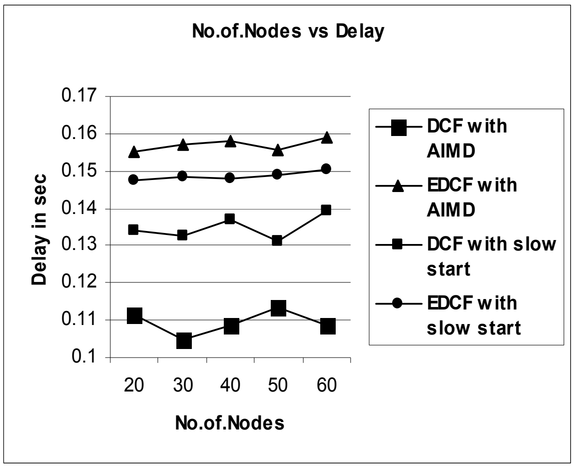

Figure8: No. of Nodes vs. Delay

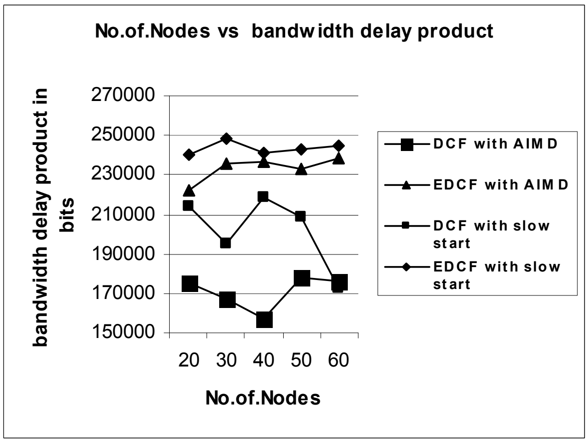

Figure.9:No.of.Nodes vs. packet loss

J.Premalatha received the B.E.degree in Electronics and Communication Engineering from Madurai Kamarajar University, and M.E in Computer Science and Engineering from Bharathidasan University, in 1989 and 2001, respectively. She is currently doing her Ph.D degree in Anna University, Chennai, India. Currently she is a assistant professor in the department of computer science and Engineering, Perundurai, Tamilnadu. Her area of interest is Adhoc Networks, Computer Networks, Artificial Intelligence and Network security. She has published articles in International Journals and presented many papers in national and International Conferences..

Dr.P.Balasubramanie has been awarded Junior Research Fellowship (JRF) by CSIR in the year 1990. He has completed his Ph.D degree in theoretical computer science in the year 1996 in Anna University Chennai. Currently he is a professor in the department of Computer science and Engineering in Kongu Engineering College, Perundurai, Tamil Nadu , India. He has guided 3 Ph.D scholars and guiding 20 scholars Under Anna Uniuversity. He has published more Than 50 articles in International/ National Journals. He has autored 6 books with the reputed publishers.

Table2: Comparison of Various QoS parameters

S.No	Parameters	No Of Nodes	802.11 With AIMD	802.11e With AIMD	802.11 With Slow Start	802.11e With Slow Start
1	Throughput (bps)	20	6640	9009	8406	9234
		30	6563	9224	8266	9337
		40	6689	9165	8366	9252
		50	6893	9213	8196	9311
		60	6794	9346	8286	9412
2	Delay(sec)	20	0.1114	0.1552	0.134	0.1477
		30	0.1049	0.1571	0.1326	0.1486
		40	0.1088	0.1578	0.137	0.148
		50	0.1134	0.1554	0.131	0.1489
		60	0.1086	0.1591	0.1391	0.1505
3	Packet Delivery Ratio (pkts)	20	255	302	290	317
		30	247	295	290	314
		40	252	300	285	316
		50	250	297	287	312
		60	256	307	290	317
4	Bandwidth Delay Product (bits)	20	175340.95	222086.24	214187.73	240017.87
		30	167028.86	235317.30	195114.45	248245.3
		40	157367.08	236575.59	218268.71	240956.52
		50	178242.20	232828.51	208455.6	242702.57
		60	176251.11	238604.65	172308.41	244467.13
5	Packet Loss	20	51	41	60	33
		30	51	41	60	36
		40	53	42	65	34
		50	54	44	63	38
		60	55	41	60	33

www.icgst.com

Design of an Efficient QoS Architecture (DEQA) for Mobile Ad hoc Networks

R.Mynuddin Sulthani[1] and D.Sreenivasa Rao [2],
[1] Professor in CSE Dept, Madanapalle Institute of Technology & Science (MITS), Madanapalle, Chittoor (Dt), A.P., India. [2] Professors in ECE Dept, JNTU College of Engg, JNT University, Hyderabad, A.P., India.
rmsulthan@yahoo.co.in, dsraoece@yahoo.co.uk

Abstract
With the increasing widespread use of wireless technologies, Quality of service (QoS) provisioning in ad hoc networks remains a challenging task. Good scalability of the QoS architecture is necessary, as the size of the ad hoc networks is huge. In this paper we propose a design of an efficient QoS architecture (DEQA) for mobile ad-hoc networks, which consists of a multi-path routing protocol, a call admission control scheme (CAC) and a congestion control mechanism. The multi-path routing protocol utilizes erasure code techniques for producing replicated fragments for each packet, to enhance reliability. Important fragments can be sent through the paths with strong connectivity and high delivery probability. While it receives an assured number of fragments, destination can recover the original packet. In the CAC scheme, requests are admitted based on the bandwidth availability of the path. To avoid performance degradation due to mobility-triggered congestion, a congestion control mechanism has been developed. Once congestion occurs, the best effort traffic is rate controlled, to free bandwidth for the real-time flows. All these schemes together make the proposed QoS architecture scalable to large network size and mobility. By simulation results, it has been shown that, this architecture allows us to provide reliable QoS in ad-hoc networks increases the available bandwidth, performs load balancing in the network and increases lifetime.

Keywords: *Scalable, Congestion, Reliable, Fragments, Bandwidth, MultiPath*

1. Introduction
The ad-hoc wireless network is recognized as an extraordinary case of wireless network, which has no fixed backbone infrastructure. Owing to the above mentioned property, wireless ad-hoc networks can be flexible and rapidly deployed, but also poses significant technical challenges. Numerous challenges include effective routing, medium access, power management, security and quality of service (QoS) issues. As the nodes communicate over wireless links, each node should fight against the highly erratic nature of wireless channels as well as interference from other transmitting nodes. These factors make it a challenging problem to maximize data throughput while meeting user-required QoS in wireless ad-hoc networks

Owing to mobility of the nodes and interference among nodes, great difficulty in implementing ad-hoc networks comes from frequent route changes. The high packet loss rates and frequent topological changes make the transport layer unstable and limit the amount of traffic that the network can carry. Three well-known problems in ad-hoc networks are the lack of reliable packet delivery due to interference and movement of nodes, limited bandwidth due to channel restrictions, and limited node lifetime due to small battery size [14].

Towards enlarging the life span of the network fairness of access to the network, research in QoS issues in wireless ad-hoc network is generally oriented. Along with it, the issues of connections among heterogeneous nodes are treated. For traffic flows in mobile ad-hoc networks, there is much need of research in offering reliable transport services as well as giving QoS guarantees.

Since the size of the ad-hoc networks is huge, good scalability of the QoS architecture is necessary. In the case of preceding works on QoS in ad-hoc networks, the scalability issues are very hardly ever measured.

On Call Admission Control (CAC) [2], there are numerous research works in wireless networks. Established end-end probing-based admission control possibly will not complete the delay necessity, because the network size is large. On the other hand, network overload can only be thwarted by CAC based on existing bandwidth not including mobility. In order to permit a flow under mobility the topology may be modified. This is ensuing in modifications of traffic distribution in the network. Therefore, even if there is sufficient bandwidth at the time of flow admission, overcrowding might still take place under mobility.

This paper proposes a design of a scalable and reliable QoS architecture for mobile ad-hoc networks. It consists of

i) a multi-path routing protocol,
ii) a call admission control (CAC) scheme and
iii) a congestion control mechanism.

Towards enhancing reliability, a multi-path routing protocol, utilizing erasure code techniques for producing replicated fragments for each packet, has been proposed. After establishing the multiple paths, they can be arranged based on their strong connectivity and delivery probability. Important fragments can be sent through the stronger paths among the replicated fragments. The subsequent stronger path can be chosen from the list, if there are some unpredicted faults that happen in the path. While it receives an assured number of fragments, destination can recover the original packet. In the proposed CAC scheme, the calls based on the bandwidth availability of the path are admitted. The QoS routing to propagate bandwidth information throughout the network is adopted. An additional congestion control scheme has been developed to thwart performance degradation as a result of mobility-triggered congestion. To assign free bandwidth for the real-time flows, best effort traffic must be rate controlled once congestion occurs.

All these schemes together make the proposed QoS architecture scalable to large network size and mobility.

2. Related Work

In [1], the authors have introduced and evaluated the DLite algorithm, a novel approach to service differentiation in ad-hoc networks, which applies a fair queuing scheme with separate queues for each service class. Late packets of delay constrained classes are dropped in intermediate routers. DLite is easy to implement and requires low computational overhead. It allows for adaptive multimedia applications and permits gradual deployment.

In [2], the authors have proposed a scalable QoS architecture for ad-hoc networks. This scheme draws upon the positive aspects of both IntServ and DiffServ, and extends upon the scalable LANMAR routing protocol to support QoS and also it is capable of incorporating mobile backbone networks (MBNs) to further improve the scalability.

In [3], the authors have proposed a new QoS framework for MANETs–Adaptive Reservation and Pre-allocation Protocol (ASAP). By using two signaling messages, ASAP provides fast and efficient QoS support while maintaining adaptation flexibility and minimizing wasted reservations

In [4], the authors have performed a study on the various queuing schemes for multi-hop wireless networks and examine the fairness and throughput performance of each scheme. Each scheme offers a different degree of fairness. While relatively simple queuing schemes require less hardware and processing budget, they inevitably lack good fairness and performance. In contrast, the scheme that provides fairness requires per-flow (i.e., network-layer flow) queuing.

In [5], the authors have presented an agent based scheme for efficient management of radio resources in hybrid wireless networks. Performance of the agent based scheme is measured in terms of successful handover rate between different wireless network architectures (e.g., WLAN, Cellular), and also by the allocated bandwidth to admitted calls.

In [6], the authors have discuss a new packet scheduling models for an multi-hop wireless network, and which ensures fair allocation of basic channel service while seeking to maximize spatial reuse. The objective of the above model is to devise effective scheduling disciplines to provide packet-level QoS in terms of throughput, delay and fairness. The authors demonstrate a packetized algorithm that realizes the scheduling model with analytically provable performance bounds. In addition a backoff-based distributed implementation is designed in the above mentioned paper, which closely emulates the ideal centralized algorithm. The above mentioned paper also demonstrates the effectiveness of the devised algorithm through both simulations and analysis.

In [7], the authors have proposed a QoS-aware routing protocol that incorporates an admission control scheme and a feedback scheme to meet the QoS requirements of real-time applications. The novel part of the above mentioned QoS-aware routing protocol is the use of the approximate bandwidth estimation to react to network traffic. The above mentioned approach implements these schemes by using two bandwidth estimation methods to find the residual bandwidth available at each node to support new streams.

In [8], the authors have performed a study on interference-aware topology control and QoS routing in IEEE 802.11-based multi-channel wireless mesh networks with dynamic traffic. Channel assignment and routing are two basic issues in such networks. Different channel assignments can lead to different network topologies. A novel definition of co-channel interference has been presented in the above mentioned paper. Based on this concept, an effective heuristic for the minimum Interference Survivable Topology Control (INSTC) problem is formally defined and presented which seeks a channel assignment for the given network such that the induced network topology is interference-minimum among all K-connected topologies, which improves the system performance by 57% on average in terms of connection blocking ratio.

In [9], the authors have discussed an approach to support delay-sensitive multimedia applications over hybrid wireless/wired networks. By the appropriate selection of the AIMD protocol parameters, wireless resources can be efficiently utilized, flow throughput can be maximized under the constraint of the delay outage probability. Simulation results have validated the analysis, demonstrated the feasibility of the approach, and shown that the AIMD protocol can outperform the non-responsive UDP protocol when they are used to support multimedia applications over hybrid networks.

In [10], the authors have presented a users' satisfaction factor (USF) defined to quantify quality of service (QoS) for different types of services such as voice, data, and multimedia, as well as for different delay constraints. This USF not only predicts the final delivered QoS during transmission, but also take advantages of the fact that different packets can be decoded at different time in the receivers. Based on this USF, four types of scheduling schemes considering tradeoffs between system performance and individual fairness are presented. The schemes presented explore the time, channel, and multi-user diversity to guarantee quality of service and enhance the network performance.

3. Overview of the architecture

In this paper, we propose a design a scalable and reliable QoS architecture for mobile adhoc networks. It consists of

- i) a multi-path routing protocol,
- ii) a call admission control (CAC) scheme and
- iii) a congestion control mechanism.

A multi-path routing protocol, utilizing erasure code techniques for producing replicated fragments for each packet, has been proposed to enhance reliability. After establishing the multiple paths, they can be arranged based on their strong connectivity and delivery probability. Important fragments can be sent through the stronger paths among the replicated fragments. The subsequent stronger path can be chosen from the list, if there are some unpredicted fault happens in the path. While it receives an assured number of fragments, destination can recover the original packet. In our CAC scheme, we admit the calls based on the bandwidth availability of the path. We adopt the QoS routing to propagate bandwidth information throughout the network. An additional congestion control scheme has been developed to thwart performance degradation as a result of mobility-triggered congestion. To assign free bandwidth for the real-time flows, best effort traffic must be rate controlled once congestion occurs.

All these schemes together make the proposed QoS architecture scalable to large network size and mobility.

4. Multipath Reliable Protocol

A. Packet Dispersion Model

To reduce the packet loss, we propose a new packet dispersion mechanism which splits the data packets at the source, into fragments and distributes the fragments on multiple parallel paths. At the destination, packets are reassembled. In order to make this mechanism efficient, we need to use an erasure code technique based on robin dispersal algorithm [13].

The source node breaks up the packet into N fragments of size s, generates K fragments of parity and transmits the total of N+K packets to the destination. For the transmission is to be successful, the destination must receive at least N fragments in at most Tm time units.

Important fragments can be sent through the stronger paths among the replicated fragments. The subsequent stronger path can be chosen from the list, if there are some unpredicted faults that happen in the path. The next section describes the process of determining stronger paths, based on connectivity and delivery probability.

B. Nodal Delivery Probability

The data transmission is based on the parameter *Node Delivery Index* (NDI), which represents the probability that a node correctly delivers data to the destination. Let NDI_i denotes the delivery index of a node N_i. The value of NDI_i initially set to zero and updated whenever there is a message transmission or timer expiration. Whenever a node N_i transmits a data packet to another node N_K, NDI_i should be updated such that

$$NDI_i = (1-\lambda)\ NDI_i + \lambda\ NDI_k, \tag{1}$$

Where NDI_k is the NDI of node N_k, $0 < \lambda < 1$, is a constant employed to keep partial memory of historic status. If N_k is the destination, then $NDI_k = 1$, because the message is already delivered to the destination successfully. Otherwise, $NDI_k < 1$. Clearly, NDI_i is always between 0 and 1.

Each node maintains a timer t1.If there is no message transmission within an interval of δ, then the timer t1 expires. The timer expiration indicates that the node couldn't transmit any data during δ. So NDI_i should be updated as

$$NDI_i = (1-\lambda) NDI_i \tag{2}$$

So from (1) and (2), we arrive that, the node delivery index (NDI) of node N_i is updated as

$= (1-\lambda)\ NDI_i + \lambda\ NDI$, If there is a data Transmission

$= (1-\lambda)\ NDI_{i\ ,}$ If there is a Timeout

C. Reliable Data Forwarding

Our Reliable Multipath Protocol (RMP) uses an Multi Path Set (MPS) comprising node-disjoint paths, determined using the AOMDV protocol [11]. The construction of An MPS of node-disjoint paths is made through succeeding calculation of the node-disjoint, shortest in number of hops, paths, using the route discovery provided network connectivity information. While fresh connectivity information is obtained, RMP efforts to determine new paths usually either proactively or reactively, following the invocation of a route discovery.

After the determination of MPS, the source S disperses each outgoing message, adding limited redundancy to the data and dividing the resultant information into pieces, which are transmitted across the MPS routes one piece per route. Even if some of the message pieces are lost or corrupted, successful reception of M out of N pieces allows the reconstruction of the message at the destination. The ratio $T = N / M$ is termed the redundancy factor, and we denote a dispersed message with redundancy T as an (M, N)-message.

The K packets are subdivided into n non-overlapping sets with M_i packets in set i, where M_i packets are transmitted on path RI. We will denote with $m = \{M_1, M_2, ..., M_n\}^T$ the allocation vector for the connection over the n paths in the network. The probability that the packet can be reconstructed is the probability that more than M of the fragment transmissions succeeded, or equivalently that less than K transmissions failed. We can express network reliability or probability of success as:

$P_{succ} = \Pr[Number\ of\ \mathrm{Re}ceived\ Fragments \geq M]$

The efficiency of path diversification is defined as:

$$\eta = \frac{Effective\ \text{Throughput}}{\text{Actual Throughput}} = \frac{M}{M+K}$$

We ignore the overhead introduced by the rest of the network.

While transmitting dispersed data, the status of the MPS routes are continuously updated based on each node's delivery index. For this, for each path, an average node delivery index (ANDI) is calculated based on the NDI of each node in the path. For each successful or failed data, the value of ANDI of the corresponding route is increased or decreased. In case the existing route encounters some unexpected link or route failure, the algorithm selects the path with the next highest ANDI, from the list of selected disjoint paths. The advantages of using of dispersion algorithm [12] and erasure code [13] is, simplified message manipulation and reliable data transmission. At the same time, however, the optimization of erasure coding parameters is usually inaccurate because they are calculated according to the current NDI of the source node. In addition, propagating many small messages in the network may incur further processing overhead and inefficiency of bandwidth utilization. In order to avoid the above problems we describe the process of message fault-tolerance based data forwarding in the subsequent section.

D. Determining Message Delivery Probability

The Message Delivery Probability (MDP) is used to represent the amount of redundancy and to indicate the importance of a given message. The MDP is defined to be the probability that at least one copy of the message is delivered to the destination by other nodes in the network. We assume that each message copy carries a field to denote the MDP. When a message is generated, its MDP is initialized to be zero.

Let us consider a node N_i, which is transmitting k copies of a data message j. Let $MDP^j{}_i$ denote the MDP of message j in the queue of node Ni. The message transmitted to a node Nr is associated with a MDP of

$$MDP^j{}_{Nr} = 1 - (1 - MDP^j{}_i)\,(1 - \mathrm{NDI_i}) \prod_{m=1}^{k} (1 - MDP^j{}_m)$$

and the MDP of the message at node Ni *is* updated as

$$MDP^j{}_i = 1 - (1 - MDP^j{}_i) \prod_{m=1}^{k} (1 - \mathrm{MDP^j}{}_m)$$

The above process repeats at each time when message *j* is transmitted to another node. In general, the more times a message has been forwarded, the more copies of the message are created, thus increasing its delivery index. As a result, it is associated with a larger MDP. So the messages with larger MDP are dispersed and sent through the best paths with high ANDI.

5. Call Admission Control (CAC)

Apart from the multipath reliable routing protocol, our QoS architecture provides Call admission Control (CAC).

When a new request with certain bandwidth requirement comes, the source will perform admission control following the steps described below [2].

The source node first consults the local routing table. If there is enough bandwidth to the destination, the source node further checks required bandwidth.

- If it is less than the minimum bandwidth, the flow can be admitted.

- If it is more than the maximum bandwidth, the flow is rejected.
- If it is between the minimum and maximum bandwidth, a probing packet is sent to the destination node to obtain the exact available bandwidth at the destination. Based on the results, it there is available bandwidth, the flow is accepted; otherwise it is rejected.

6. Congestion Control

In mobile ad-hoc networks, call admission control at source nodes alone cannot guarantee QoS since the topology may change after flows are admitted. Network congestion can still occur frequently under mobility. Thus, congestion control is needed to provide QoS in such situations. When congestion occurs, we would like best-effort traffic to first reduce their transmissions rate to give bandwidth to real-time flows.

A. Network Congestion Detection

For congestion control, we need to first detect the congestion in the network. In multi-hop wireless networks, accurate congestion detection of a neighborhood is difficult. In this paper, congestion detection in a node's neighborhood is performed by monitoring the wireless channel utilization ratio, which can be obtained by the adaptive bandwidth estimation scheme. When the channel utilization ratio is larger than a pre-defined threshold value, it can be assumed that this node's neighborhood is entering a congested state.

B. Rate Control of Best Effort Traffic

By applying rate control to best-effort traffic, congestion control can be performed. The best-effort traffic may use unconsumed bandwidth of the real-time flows. But whenever a new real-time flow arrives, the best effort traffic flows are forced to free the bandwidth for the new real-time flow. The starvation of best effort flows can be avoided by reserving a small fraction of the bandwidth to best-effort traffic at all times. The basic idea of the scheme is, when there is no congestion, the best-effort traffic increases its rate slowly. Once congestion is detected, the ECN bits of best-effort traffic will be marked and transmitted to the source nodes. On receiving this ECN packet, source nodes will reduce their sending rate.

7. Experimental Results

A. Simulation Model and Parameters

We use Network Simulator (NS2) to simulate our proposed algorithm. In our simulation, the channel capacity of mobile hosts is set to the same value: 2 Mbps. We use the distributed coordination function (DCF) of IEEE 802.11 for wireless LANs as the MAC layer protocol. It has the functionality to notify the network layer about link breakage.

In our simulation, 50 mobile nodes move in a 1000 meter x 1000 meter rectangular region for 50 seconds simulation time. We assume each node moves independently with the same average speed. All nodes have the same transmission range of 250 meters. In this mobility model, a node randomly selects a destination from the physical terrain. It moves in the direction of the destination in a speed uniformly chosen between the minimal speed and maximal speed. After it reaches its destination, the node stays there for a pause time and then moves again. In our simulation, the minimal speed is 5 m/s and maximal speed is 10 m/s. The simulated traffic is Constant Bit Rate (CBR).

Our simulation settings and parameters are summarized in table 1

No. of Nodes	50
Area Size	1000m X 1000m
Mac	IEEE 802.11
Radio Range	250m
Simulation Time	50 sec
Traffic Source	CBR
Packet Size	512
Speed	10m/s
Flows	2,4,6,8 and 10
Rate	250,500,750, & 1000 Kb

Table 1: Simulation Parameters

B. Performance Metrics

The proposed DEQA is compared with the SMT [13] scheme**.** The performance is evaluated mainly, according to the following metrics.

i. **Average end-to-end delay:** The end-to-end-delay is averaged over all surviving data packets from the sources to the destinations.
ii. **Average Packet Delivery Fraction:** It is the ratio of the number .of packets received successfully and the total number of packets transmitted.
iii. **Aggregated Throughput:** We measure aggregated throughput in terms of no. packets received.
iv. **Fairness:** For each CBR flow, we measure the fairness as the ratio of throughput of each flow and total no. of flows.
v. **Packet Loss:** We measure the packet loss , which is the no. of packets lost per unit time.
vi. **Blocking Probability:** We measure the blocking probability as the ratio of rejected requests per total no. of requests.
vii. **Overhead:** It is the overhead occurred due to control packets exchange.

C. Results

In this section simulation results are presented.

1. Rate

In the first experiment, the transmission rate is varied as 250kb; 500kb…1000Kb and the above metrics are measured.

Fig1 shows the result of throughput in terms of packets. From the figure, it can be seen that DEQA scheme outperforms SMT, by gaining more packets.

Fig 2 shows the PDF for DEQA and SMT. Clearly DEQA's PDF is significantly more than that of SMT.

Fig 3 gives the end-to-end delay values. Due to its routing policy, DEQA has less delay when compared to SMT.

Fig. 4 presents the packets dropped for both the schemes. Because of its reliability, DEQA has less packet drops than SMT.

The blocking probability results for various bandwidth requests are presented in figure 5. Because of its call admission control policy, DEQA has reduced blocking probability, when compared with SMT.

The fairness of all the flows are given in figure 6. Since the DEQA scheme allocates fair bandwidth for its flows, it has increased fairness when compared to SMT.

Fig 7 shows that the Overhead required for SRQA is less compared overhead of SMT.

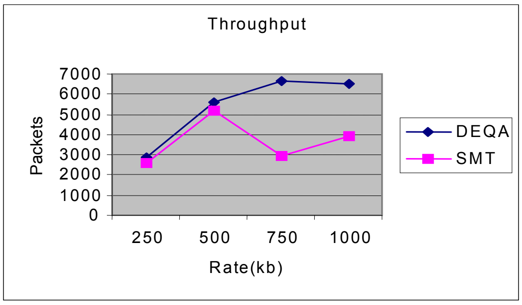

Fig.1 Rate Vs Throughput

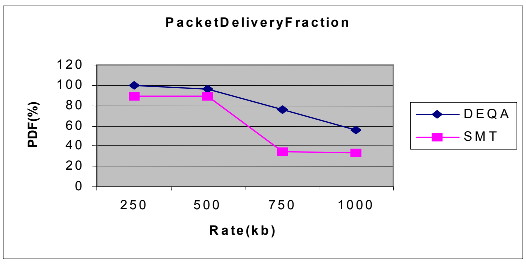

Fig.2 Rate Vs PDF

2. Flows

In the second experiment, the no. of data flows is varied as 2, 4, 6,8,10 and the same metrics are measured.

Fig8 shows the result of throughput in terms of packets. From the figure it can be seen that, DEQA scheme outperforms SMT, by gaining more packets.

Fig 9 shows the PDF for DEQA and SMT. Clearly DEQA's PDF is significantly more than that of SMT.

Fig 10 gives the end-to-end delay values. Due to its CAC and congestion control schemes, DEQA has less delay when compared to SMT.

Fig. 11 presents the packets dropped for both the schemes. Because of its reliability, DEQA has less packet drops than SMT.

The blocking probability results for various bandwidth requests are presented in figure 12. Because of its call admission control policy, DEQA has reduced blocking probability, when compared with SMT.

The fairness of all the flows are given in figure 13. Since the DEQA scheme allocates fair bandwidth for its flows, it has increased fairness when compared to SMT.

Fig 14 shows that the Overhead required for SRQA is less compared overhead of SMT.

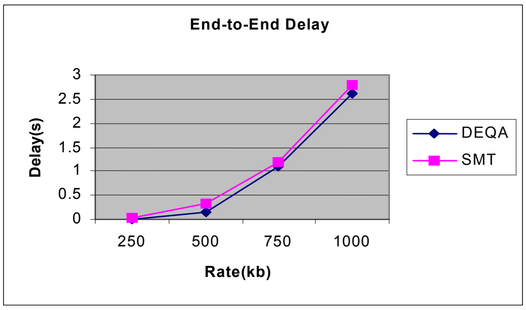

Fig.3 Rate Vs Delay

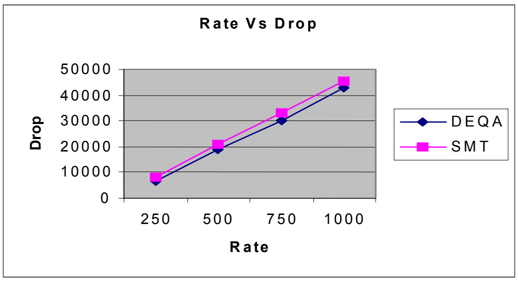

Fig.4 Rate Vs Packets Dropped

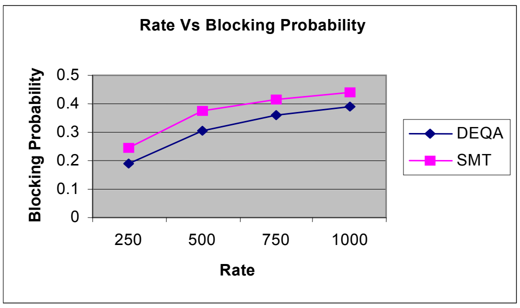

Fig.5 Rate Vs Blocking Probability

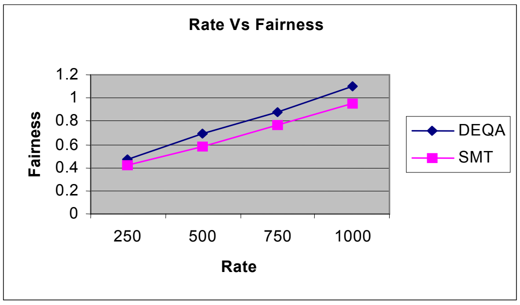

Fig.6 Rate Vs Fairness

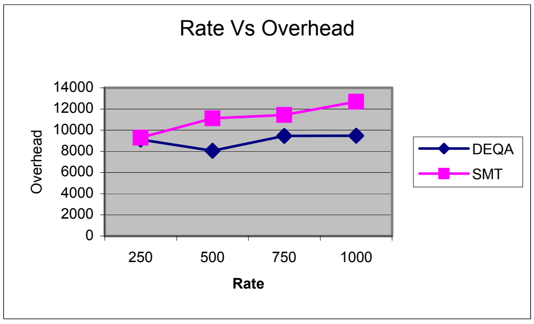

Fig.7 Rate Vs Overhead

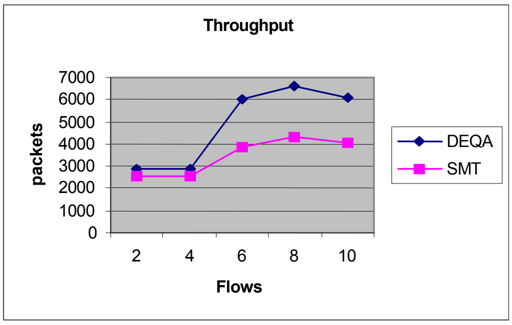

Fig.8 No. of Flows Vs Throughput

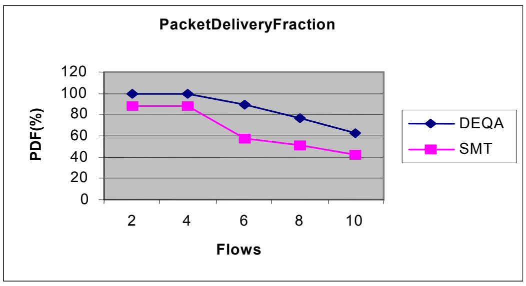

Fig.9 No. of Flows Vs PDF

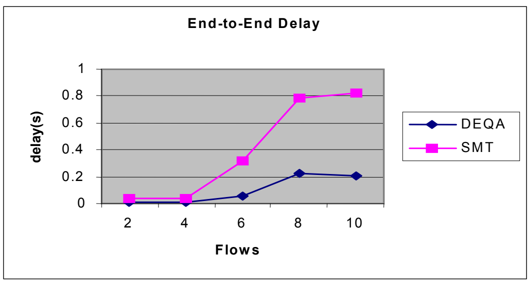

Fig.10 No. of Flows Vs Delay

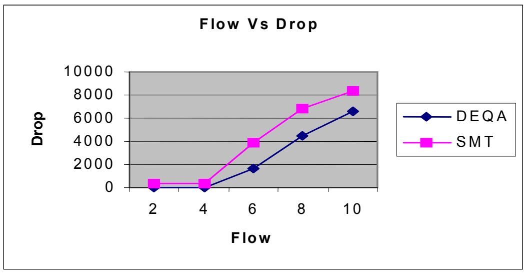

Fig.11 No. of Flows Vs Packets Dropped

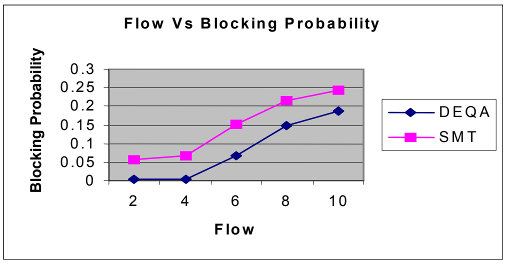

Fig.12 No. of Flows Vs Blocking Probability

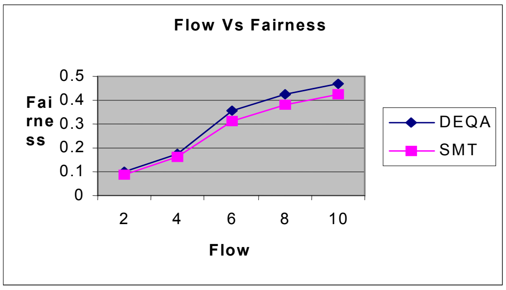

Fig.13 No. of Flows Vs Fairness

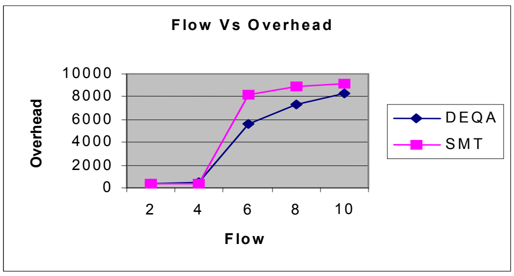

Fig.14 No. of Flows Vs Overhead

9. Conclusion

In this paper, a joint design of reliable QoS architecture for mobile adhoc networks has been designed which consist of a multipath routing protocol, a call admission control scheme and a congestion control mechanism. In the reliable multi-path routing protocol, dispersion and erasure code techniques are utilized for producing replicated fragments for each packet, to enhance reliability. Then messages with good delivery probability are identified and transmitted through the paths with high average node delivery index. While it receives an assured number of fragments, destination can recover the original packet. Next, a call admission control (CAC) scheme has been developed, in which, the calls are admitted based on the bandwidth availability of the path. Finally, a congestion control mechanism has been developed to avoid performance degradation due to mobility-triggered congestion. Once congestion occurs, the best effort traffic is rate controlled, to free bandwidth for the real-time flows. By simulation results, it has been shown that, this combined QoS architecture achieves good throughput, reduces packet loss, increases available bandwidth, performs load balancing in the network and increases lifetime.

References

[1] Michael Gerharz, Christian de Waal, Matthias Frank, "A Practical View on Quality-of-Service Support in Wireless Ad Hoc Networks". In Proc. of the 3rd IEEE Workshop on Applications and Services in Wireless Communications and Mobile Computing 2002.

[2] K. Xu, K. Tang, R. Bagrodia, M. Gerla, and M. Bereschinsky, “Adaptive Bandwidth Management and QoS Provisioning in Large Scale Ad Hoc Networks,” Proc. Military Comm. Conf. (MILCOM '03), Oct. 2003

[3] Xue, J. Stuedi, P. Alonso, G.,"ASAP: an adaptive QoS protocol for mobile ad hoc networks",14th IEEE proceedings on Personal, Indoor and Mobile Radio Communications, 2003. PIMRC 2003.

[4] Jangeun Jun Sichitiu, M.L.,"Fairness and QoS in multihop wireless networks", IEEE 58th Vehicular Technology Conference, 2003. VTC 2003-Fall.

[5] R. Babbar, A.O. Fapojuwo, B. Far, “Agent-based resource. Management in hybrid wireless networks,” accepted for presentation, IEEE Canadian Conference on Electrical and Computer Engineering, 2004

[6] Haiyun Luo, Songwu Lu, Vaduvur Bharghavany, Jerry Cheng, Gary Zhongy, "A Packet Scheduling Approach to QoS Support in Multihop Wireless Networks", ACM Journal of Mobile Networks and Applications (MONET) Vol. 9, No. 3, June 2004

[7] Lei Chen,Wendi B. Heinzelman, "QoS-Aware Routing Based on Bandwidth Estimation for Mobile Ad Hoc Networks", IEEE Journal on selected Areas in Communications, vol. 23, no. 3, March 2005.

[8] J. Tang, G. Xue, and W. Zhang, "Interference-aware topology control and QoS routing in multi-channel wireless mesh networks,", Proceedings of the 6th ACM international symposium on Mobile ad hoc networking and computing, May 25–27, 2005.

[9] Lin Cai, Xuemin (Sherman) Shen, Jon W. Mark, and Jianping Pan, "QoS Support in Wireless/Wired Networks Using the TCP-Friendly AIMD Protocol", IEEE Transactions On Wireless Communications, VOL. 5, NO. 2, FEBRUARY 2006.

[10] Zhu Han; Xin Liu; Z. Jane Wang; K. J. R. Liu, "Delay sensitive scheduling schemes for heterogeneous QoS over wireless networks", IEEE Transactions on Wireless Communications, Volume 6, Issue 2, Feb. 2007 Page(s):423 – 428

[11] Mahesh K.Marina, Samir R.Das “Ad hoc On-demand Multipath Distance Vector Routing”, ACM Sigmobile Mobile Computing and Communications Review Volume 6 , Issue 3, July 2002.

[12] Panagiotis Papadimitratos, Zygmunt J. Haas, “Secure Data Communication in Mobile Ad Hoc Networks”, IEEE Journal On Selected Areas In Communications, Vol. 24, No. 2, February 2006.

[13] Panagiotis Papadimitratos, Zygmunt J. Haas “Secure message transmission in mobile ad hoc networks ”, 2003 Elsevier, Ad Hoc Networks 1 (2003) 193–209.

[14] P. Djukic, "Optimum Resource Allocation in Multipath Ad Hoc Networks", MASc Thesis, University of Toronto, 2003

R.Mynuddin Sulthani received the B.E., degree in Electronics and Communication Engineering from Osmania University, Hyderabad, India in 1992, and M.Tech., in Digital Systems & Computer Electronics from JNT University, Hyderabad, India. He is pursuing his PhD work on Quality of Service (QoS) in wireless Networks, under JNT University, Hyderabad, India. His research interests are in the areas of wireless/mobile networks with emphasis on the QoS guarantee and adaptation, connection and mobility management, and wireless MAC protocols. He is currently working as a Professor & HOD in CSE Department, Madanapalle Institute of Technology & Science (MITS), Madanapalle, which is affiliated to JNTU, Hyderabad, India. He has 7 years of Industrial experience and 10 years of teaching experience. He has attended an international conference ICCCN-2008 and several Workshops conducted by various organizations in India. He has organized national level seminars, workshops and refresher courses in MITS.

D.Sreenivasa Rao received the B.Tech., degree in Electronics and Communication Engineering in 1986 from Nagarjuna University, India and M.E in Digital Systems in 1994 from Osmania University, Hyderabad, India. He was awarded the PhD degree in Computer Networks in 2004 from University of Hyderabad, India. He is a Professor in ECE Department, JNTU College Of Engineering, JNT University, Hyderabad, India. He has 20 years of teaching experience. His research interest is in the area of communications and computer networks Presently 12 research students are working under his guidance. He has 22 publications in various National, International Conferences and Journals. He has 20 years of teaching experience. He has attended more than 10 Short Term Courses, Summer Schools, and Workshops, conducted by various organizations. He has organized workshops and refresher courses. He has chaired sessions at various national conferences. He is advisory committee member for GNIT, Hyderabad. He is also governing body member for Syed Hashim College of Science & Technology, member of JNTU forum for Science & Society and Coordinator for campus networking at JNTU CE, Hyderabad